Cisco IP Communications Express: CallManager Express with Cisco Unity Express

By Danelle Au, Baldwin Choi, Rajesh Haridas, Christina Hattingh, Ravi Koulagi, Mike Tasker, Lillian Xia

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Overview

Master your branch or small office call routing with CallManager Express IP Telephony tips from the product team at Cisco Systems Learn how to get the most out of your Cisco CallManager Express (CCME) system Gain insights and tips from the Cisco product team that are not available in product documentation Control branch office and small business call routing with maximized use of CCME features Create unique solutions from vertical market application examplesCisco CallManager Express (CCME) is an IOS-based call control engine for IP Phones and PSTN Gateway calls integrated into Cisco branch office router products. It is part of the Cisco AVVID portfolio, addressing the small- and branch-office end of the market of 120 or fewer people per office. These can be standalone offices or enterprise branch offices networked together via IP, typically too small to make Cisco CallManager with a Cisco Unity messaging system cost-effective to purchase or maintain.Cisco CallManager Express provides CCME tips and guidance not available in product literature or from other books. The book shows how the product features can be deployed together and customized to provide specific fits for different vertical market segment needs. The book is organized in three main sections: market segment overview and definition; CCME capabilities: services, operation, design, deployment, applications, networking, and customer scenarios for specific vertical segments; and maintenance and troubleshooting.

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Cisco IP Communications Express:

CallManager Express with Cisco Unity Express

Danelle Au, Baldwin Choi, Rajesh Haridas, Christina Hattingh, Ravi Koulagi, Mike Tasker, Lillian Xia

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Christina Hattingh is a member of the technical staff in the branch office IP Communications group at Cisco Systems. The products in this group, including the Cisco 2600, 2800, 3600, 3700, and 3800 series platforms, were some of the first Cisco platforms to converge voice and data by offering Public Switched Telephone Network (PSTN), and private branch exchange (PBX) voice interfaces and critical QoS features on WAN interfaces. More recently, these products have integrated call control elements such as Cisco CME and Cisco UE into the router-based platform. In this role, Hattingh helps guide development projects, trains Cisco sales staff and Cisco resale partners on new router-based voice technologies, and advises customers on voice network deployment and design.

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Acknowledgments

Writing this book was a team effort because timing, availability, and accuracy were critical. In the fast-moving industry of voice over IP (VoIP) technology, information and products age quickly. A single author could not have produced this book in the time needed, nor does any one person know all the aspects of the system to the depth required to write all the chapters of this book. The seven authors of this book cover diverse aspects of the system in their daily work, including defining requirements, training the sales force and customers, designing and implementing the software, and testing and certifying system operation.

We would like to thank Sarat Khilnani, Tomoo Esaka, and Markus Schneider for reviewing this book and suggesting countless improvements. The larger Cisco product teams were invaluable in their support of the author team and making knowledge and equipment available toward the production of this bookin particular, Ed Leonhardt, Andy Feest, Dean Galanos, Praveen Sankaran, Vallinath Panchagnula, Subodh Shah, Narendra Hosehalli, Chandrodaya Prasad, Yuan Cai, and Haitao Zhang. Our thanks also to the Cisco management team for supporting us in this endeavor, including Michael Wood, Deependra Vaidya, Jiabin Zhao, Sarat Khilnani, and Sam Lyall. Thank you also to Tim Redpath of Metavante, who helped with questions on implementations in different industries.

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Icons Used in This Book

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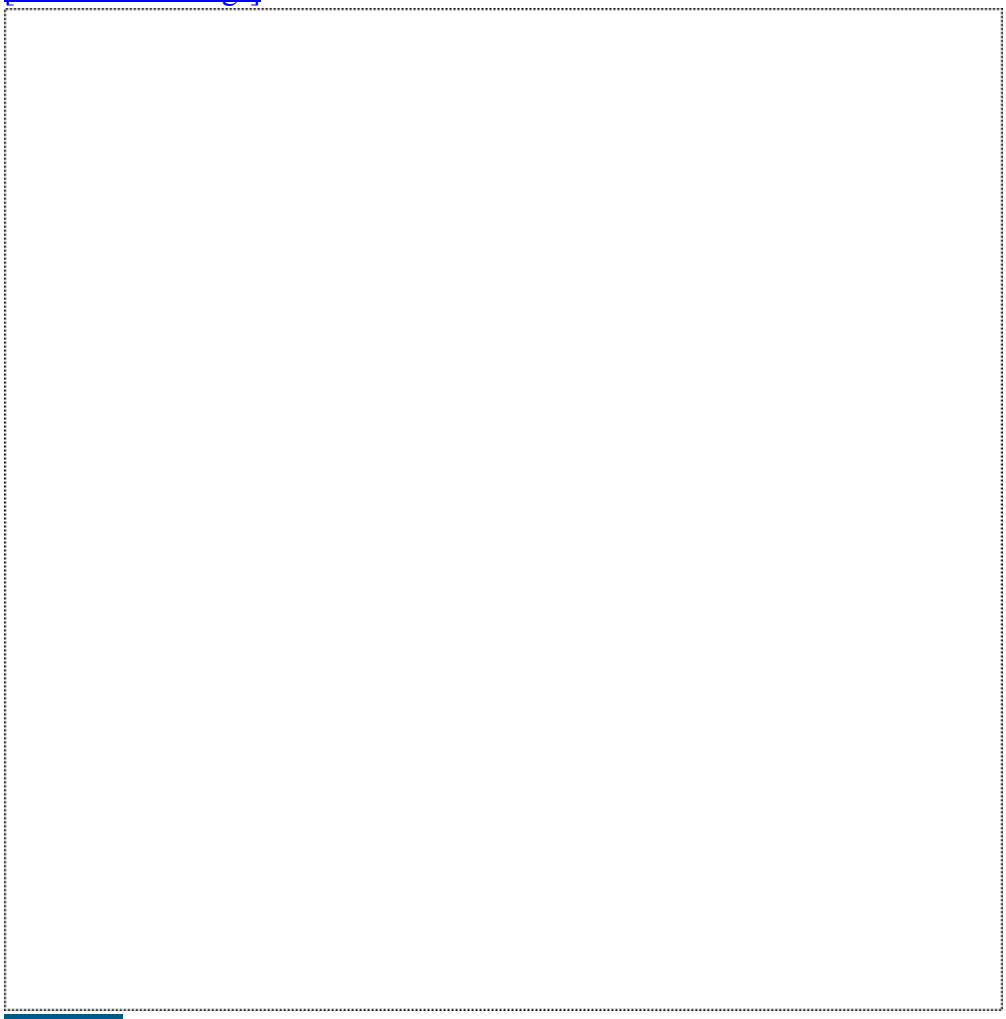


Command Syntax Conventions

The conventions used to present command syntax in this book are the same conventions used in the IOS Command Reference. The Command Reference describes these conventions as follows:

- Bold indicates commands and keywords that are entered literally as shown.
- Italic indicates arguments for which you supply actual values.
- Vertical bars (|) separate alternative, mutually exclusive elements.
- Square brackets ([ ]) indicate an optional element.
- Braces ({ }) indicate a required choice.
- Braces within brackets ([{ }]) indicate a required choice within an optional element.

Introduction

Cisco IP Communications Express: CallManager Express with Cisco Unity Express is the first comprehensive book describing Cisco converged communications solutions for the small standalone or enterprise branch office. Much has been written on solutions for larger enterprise networks, but a dearth of information exists for solutions for offices employing fewer than 100 people.

IP-based communications solutions for the small or branch office are now a reality in terms of technology, price, and product availability. Advanced features in an easy-to-use, single-box packaging now offer exciting new options for the small-business owner, or for smaller sites of a larger network. Wherever a router is deployed to provide data traffic or Internet access, that platform can also be leveraged or upgraded to provide full telephony features and data access for the office.

The small office requires a few PSTN trunks, phone handling with typical office features such as transfer and conference, an automated attendant (AA) to handle general or after-hours calls, voice mail, security features such as a firewall, and Internet access for e-mail and website applications. All this is now provided in a single Cisco router-based platform for offices of up to 240 employees.

Goals and Methods

This book tells you how to use a Cisco IPC Express system to take your business a leap forward. Perhaps you are setting up a new office or business, replacing an aging solution, wanting to move into the IP world to gain the productivity benefits of converged networks for your employees or colleagues, or starting a VoIP pilot operation as part of a larger network. Whatever the case, this book tells you how the small-office IP communications system works, how you can configure its rich set of features to meet your needs, your choices in administering the system, the features and benefits it offers, and how to isolate and correct any problems you might encounter.

For the system integrator or value-added reseller, this book arms you with the skills to offer Cisco IPC Express to your customers, to set it up quickly and efficiently, to understand its features and operation, and to save time in deploying a leading-edge VoIP system for a business owner or customer network. This book also helps you prepare for Cisco IPC Express certification.

Who Should Read This Book?

This book helps you understand the Cisco IPC Express system and its operation, regardless of whether you are the system's current or prospective owner or administrator or whether you are in the business of reselling communications systems to small-business owners.

If you own a small business and are looking for a new or upgraded communications system, this book helps you evaluate your options and prepare for your conversations with the reseller of your choice. If you already own the system, this book helps you configure, maintain, and troubleshoot it.

If you are part of the IT infrastructure of a larger enterprise, this book helps you understand how you can quickly and easily start a VoIP pilot in select offices of your organization without needing the immediate commitment or planning to do a full-scale migration to IP telephony everywhere. With Cisco IPC Express you can start small, and migrate your network at your own speed and within your budget.

If you are a reseller, Cisco IPC Express is an exciting product to add to your suite of offerings, and this book helps you prepare to sell, install, configure, and support it. If you have ideas for value-added services and applications you may offer to your customers, this book tells you how to integrate these with the system.

How This Book Is Organized

Although you can read this book cover to cover, it is designed in four main parts as different areas of interest. You can focus on the topics that are most relevant to your business, or you can progress through the chapters in sequence, learning the fundamentals before tackling more complex topics. This is particularly important if you want to become Cisco certified in reselling the system or want to be proficient in understanding its operation and features.

The following describes the parts and chapters that make up this book:

Part I, "Cisco IP Communications Express Overview," introduces Cisco IPC Express and discusses where it fits into the Cisco IP telephony portfolio. Read these chapters if you are unfamiliar with Cisco IPC Express and want to get an overview of what the system can do for you or your business, or if you are shopping for a small-office IP-based communications solution and are curious about what the industry has to offer. Also read this part of the book to see how Cisco IPC Express can benefit your business, how it fits into your network design, and to get an overview of the system architecture.

- Chapter 1, "Introducing Cisco IPC Express," provides an overview of Cisco IPC Express, describes the system's components, and reviews how it fits into the overall Cisco IP telephony product portfolio. This chapter also explains the benefits and cost of ownership of Cisco IPC Express, as well as how it can be deployed in certain industries.
- Chapter 2, "Building a Cisco IPC Express Network," discusses Cisco IPC Express network deployment options. It briefly covers general IP telephony network types and then covers the details of single-site and multisite network design considerations particular to Cisco IPC Express. This chapter explores networks in which every site has a Cisco IPC Express system, as well as hybrid networks that also contain sites under the control of a Cisco CallManager system.
- Chapter 3, "Cisco IPC Express Architecture Overview," provides an overview of the Cisco CME and Cisco UE components of the Cisco IPC Express system. It explains the Cisco IOS software structure and foundation that Cisco CME is built on and provides background understanding for the feature operation discussion in Part II. The hardware and software architecture of Cisco UE is also explained, providing a foundation for understanding how its applications interact with Cisco CME and how the components work together to provide an integrated IP telephony system for your office.

Part II, "Feature Operation and Applications," contains detailed discussions of feature operation and configuration considerations to help you deploy Cisco IPC Express to its best advantage to solve the business problems you or your customers are facing. This part of the book is also a helpful reference if you're preparing for the Cisco CME certification exam, because it discusses in depth how the system works and interacts with surrounding applications.

- Chapter 4, "Cisco IP Phone Options," provides details on the Cisco IP phones that are supported and the phone features you can use with Cisco CME.
- Chapter 5, "Cisco CME Call Processing Features," discusses Cisco CME features such as PBX and key system configurations, shared-line appearances, hunt groups, ephone-dns, dial peers, overlay directory numbers (DNs), intercom, paging, call pickup, call park, transfer, forwarding, softkey customization, and billing considerations.
- Chapter 6, "Cisco CME PSTN Connectivity Options," provides background information on connecting your Cisco CME system to the PSTN. It covers concepts such as analog and digital signaling on PSTN trunks, direct inward dial (DID), dialed number identification service (DNIS), and calling line ID (CLID). The

What You Will Find on the Website

A simple Cisco UEs AA is used in Part IV of this book to illustrate how to isolate potential issues with AA scripting. Chapter 20 discusses the three script files that comprise this sample AA, and Appendix B shows a text representation to explain the script flow. You can download the binary script files from the website to your computer by going to http://www.ciscopress.com/title/158705180X and opening the files with the Cisco Unity Express Script Editor. (You can download the Editor application from Cisco.com.)

Part I: Cisco IP Communications Express Overview

Chapter 1: Introducing Cisco IPC Express

Chapter 2: Building a Cisco IPC Express Network

Chapter 3: Cisco IPC Express Architecture Overview

Chapter 1. Introducing Cisco IPC Express

This chapter covers the following topics:

- The benefits of Cisco IP Communications (IPC) Express to your business
- The components of the Cisco IPC Express system
- Using IP Communications in your enterprise branch or small- or medium-sized business office
- How Cisco IPC Express fits into the overall Cisco IP Telephony product portfolio

It is the summer sale season at the clothing store where you work, and the floor is swamped with shoppers. One customer inquires about an item that is unavailable. With a few keystrokes on your IP phone, you check the inventory database to see if there is a similar item in another store location. As you input the customer's account number on the phone to reserve the item, the database informs you that this person is one of your best customers. You pick up the IP phone handset and page an employee to provide special assistance to the customer. The employee intercoms you using his wireless IP phone from another section of the store and promises to head over.

When you finish assisting the customer, your IP phone message waiting indicator (MWI) light comes on. You press a button on the IP phone and scroll through your messages. You have five voice mail messages and two e-mail messages. You return an urgent message from your supplier by pressing a button on your phone before attending to the less urgent messages.

Welcome to the world of convergence, where data and voice are delivered on a single IP network through IP communications systems. The preceding example only touches on the promise of IP telephony. There are many more reasons why it can transform your business, whether you are in the retail, financial, healthcare, or any other industry. First, IP telephony delivers converged applications in a cost-effective manner to users who may not have had access to them before. Instead of providing every employee in an office with a powerful computer, an IP phone can serve as the business communications device for many employees. Second, regardless of our collective readiness, the world of convergence has arrived. Just like the Internet revolution caught some by surprise, businesses that are slow to migrate will find themselves vulnerable to competitors that provide better services and have more productive employees. Finally, from a return-on-investment perspective, IP telephony reduces the complexity of having to manage and maintain separate network infrastructures for data and voice.

This book focuses on an IP telephony system called Cisco IP Communications (IPC) Express. Cisco IPC Express is a business communications solution that offers a comprehensive set of telephony features, as well as integrated routing, security, Ethernet switching, and numerous other applications on a single platform. The primary components of the Cisco IPC Express system are

- The call processing software Cisco CallManager Express (CME)
- The IP Communications platform, ranging from the Cisco 1700 to the Cisco 3800 series Integrated Services Routers (ISRs), supporting the Cisco IOS software that provides routing, security, and numerous networking interfaces
- The IP-based applications (such as Cisco Unity Express (UE) and Cisco Unity), offering automated attendant (AA), voice mail, and various phone-based services

The Purpose of Cisco IPC Express

In an IP telephony network, voice and data traffic converge from their separate sources onto a single IP network infrastructure. Most communications solutions available today, however, still deploy multiple platforms for different functions in the network. For instance, a private branch exchange (PBX) or key system delivers the telephony infrastructure, and a separate LAN switch delivers local data connectivity. A router manages Internet or WAN access, and a virtual private network (VPN) appliance, intrusion detection system, and firewall deliver the security infrastructure. As a result, the financial investment and management resources to deploy a multiple-platform network add up quickly.

The Cisco IPC Express system integrates data, voice, security, and routing into a single IP communications platform. No separate server or appliance is required for any of these network capabilities. You can also integrate additional features, such as content delivery, web application caching, URL filtering, LAN switching, wireless device access, network access control (NAC), and intrusion detection to protect your office from virus and worm infections into the same platform. In the Cisco IPC Express solution, all these applications are modular and can be added or turned on optionally based on what you need in your business. This means that businesses can deliver the applications that meet their initial requirements and then add other applications and functions as required when the business expands.

In addition, with data communications now a ubiquitous service in any business, adding telephony services to the existing data infrastructure is one way to protect your investment in equipment. A majority of enterprises and small or medium businesses have invested in a robust data infrastructure to communicate effectively with their vendors and customers. A business with a data infrastructure using the Cisco 2600XM, 2800, 3700, or 3800 ISRs can use these same platforms to deploy Cisco IPC Express.

Because the data infrastructure deployed by businesses today has been fine-tuned over the years to deliver reliable features, the expectation is that a telephony solution deployed on the data infrastructure will provide the same or better system reliability, flexibility, and availability. The telephony service is required to remain in operation, and users must continue to receive and make calls in the event of a network or power failure. Cisco IPC Express supports Hot Standby Router Protocol (HSRP) features and can be used with Survivable Remote Site Telephony (SRST) to deliver increased redundancy and availability for telephony features. Refer to the section "Other Cisco IP Telephony Solutions for the Enterprise Branch and Small and Medium Offices," later in this chapter for more details on SRST.

Together, all this provides a cost-effective, scalable, secure, and reliable communications solution that can be easily deployed and managed by both enterprise branch offices and small or medium businesses of one or more sites.

Benefits of Cisco IPC Express

Cisco IPC Express's unique hardware and software architecture, integrating all the telephony, routing, and application needs that may be required by a growing business, yields two categories of benefits: lower cost of ownership and productivity enhancements.

Lower Cost of Ownership

One of the most important aspects of the decision to acquire a new communications system is how it affects your budget, or cost of ownership. The cost includes direct costs, such as equipment costs, and hidden or indirect costs, such as ongoing maintenance or support fees. This section describes the benefits of Cisco IPC Express in lowering the total cost of ownership for businesses.

Reduce Equipment Costs

Cisco IPC Express allows enterprises to converge their voice and data network infrastructure from multiple vendors into a single IP-based network. By doing so, you save not only the costs of maintaining separate PBXs or key systems from a variety of vendors, but also the expense of maintaining a multiple-platform network with IP and non-IP traffic.

Some small or medium business owners require multiple devices for data and voice communications, such as phones, routers, switches, security appliances, and application servers. Such business owners not only have to incur the costs of acquiring the equipment, but also have to purchase maintenance contracts from different vendors and integrate all these devices and appliances into the network. When a communication breakdown occurs, it can be unclear which device is responsible and, therefore, which vendor to call for support. Cisco IPC Express's integrated architecture simplifies these deployment and maintenance activities and reduces equipment and support costs.

Reduce Upgrade Costs

By migrating from a PBX or key system to Cisco IPC Express, your business can eliminate the need to upgrade individual PBX and key systems, often from disparate vendors. PBX or key systems nearing the end of their product or lease life cycles must be replaced or substantially upgraded to enable the latest applications.

Reduce Wiring Costs

Cisco IPC Express lets you optimize a building's wiring. Both computers and IP phones use the same Ethernet port, thereby requiring only a single type of wiring in a building. In addition, only a single Ethernet port or jack is required for each user, because the computer can be plugged into the IP phone's data port. Traffic remains logically separated via the use of virtual LAN (VLAN) configurations. These ensure that data traffic does not interfere with voice traffic for purposes of quality of service (QoS) and security.

Reduce Network Administration Costs

Cisco IPC Express reduces network administration costs by more efficient and simplified network management, because there is only a single IP system to manage per site versus multiple devices from potentially different vendors.

In addition, because the challenges of network management are limited to maintaining a single system, a smaller system administration staff can manage a larger number of sites and provide faster response times. The same enterprise or small or medium businesses that previously may have outsourced a number of responsibilities to external network management contractors can now gain better control over their network and make configuration changes themselves. Doing so lowers maintenance costs and better utilizes IT resources. Training costs for staff are also reduced because of the converged system.

Reduce the Cost of Adds, Moves, and Changes

A growing enterprise or small- to medium-sized business that makes frequent additions to employee headcount or

Cisco IPC Express System Components

This section describes the components of a Cisco IPC Express system in detail. As previously explained, Cisco IPC Express includes the call processing software (Cisco CME), the IP Communications platform, the IP-based applications, and the IP-based endpoints.

Cisco CallManager Express

Cisco CME is a Cisco IOS-based call processing system that provides a wide range of IP telephony features for small- or medium-sized businesses and enterprise branch offices with up to 240 users. Cisco CME provides a cross section of traditional telephony features in addition to advanced converged features that are unavailable on most traditional telephony solutions. Businesses can also select between key system and PBX modes of operation or a combination of the two on a single network.

Because Cisco CME is completely IP-based call processing software, the software design has no physical connection constraints. This means that you can configure numerous phone and Public Switched Telephone Network (PSTN) trunk combinations on Cisco CME, delivering a range of applications for businesses. For example, the hunt group feature lets you program multiple phones in your business to have extensions that ring sequentially. In other words, when the first phone is busy or receives an incoming call that is not answered, the call rolls over to the second phone in the hunt group in sequential order, and so on. In a busy business environment, the hunt group feature or the shared-line/multiline appearance feature allows many options for implementing call coverage such that any group of employees can answer customer phone calls in the most efficient manner.

The initial installation of Cisco CME is accomplished easily using the Configuration Wizard. This setup tool prompts you for answers to a number of pertinent questions to set up system parameters. Voice mailboxes can also easily be added using the Cisco UE Initialization Wizard.

Cisco CME offers the option of Cisco IOS software command-line interface (CLI) or a web-based GUI for everyday administration and configuration. The CLI is the same interface used to configure routers, switches, and the IP Communications platforms and is familiar to most system administrators. For nontechnical staff, the web-based GUI is a simple GUI to add users, phones, and extensions or to make configuration changes.

Cisco CME is enabled with the purchase of a Cisco CME feature license with your Cisco IOS release for the router platform. Appendix A, "Cisco IPC Express Features, Releases, and Ordering Information," summarizes the Cisco CME features.

IP Communications Platforms

Cisco CME runs on the Cisco IPC Express platforms, including the Cisco 1700, 2600XM, 2691, and 3700 Access Router series, as well as the Cisco 2800 and 3800 ISR series. These communications platforms feature a diverse set of analog and digital PSTN trunk interfaces, analog station interfaces, and modular extension slots where you can add a variety of options such as integrated switching, hardware virtual private network (VPN) acceleration, voice mail, and intrusion detection systems. Depending on the number of users in the office, the right-sized IP communications platform can be selected to perform routing, switching, security, and other services.

Businesses already using access routers or ISRs for their data connectivity can easily deploy Cisco IPC Express on these platforms as well. Alternatively, businesses that have chosen to deploy IP telephony for new sites or offices can purchase a complete bundle package that includes the platform, software, licenses, and applications. Because the platforms provide such a wide range of features in a single chassis, it is easy to manage and configure.

The Cisco IPC Express platforms provide easy modularity. The interface cards used for value-added applications or connectivity can be reused across the portfolio of routers. In addition, if a business outgrows a Cisco IPC Express deployment and decides to move to a Cisco CallManager system, the equipment and licenses already purchased can be reused for this migration, preserving the entire investment.

The portfolio of Cisco IPC Express platforms offers a wide range of routing capacity, IP phone support, and PSTN

Using Cisco IPC Express in Retail, Financial, and Healthcare Businesses

This section illustrates how Cisco IPC Express is used in the retail, financial, and healthcare industries. These case studies clarify the range and types of features available with Cisco IPC Express and how they are used in different business types. Additional industries that have found Cisco IPC Express an attractive option include education (schools), government agencies, and transportation firms.

The examples discussed in the following sections are also applicable to a wide range of small and medium businesses, such as law firms, accounting firms, and real-estate agencies. Some of the differences in small and medium business solutions are in the platforms, Cisco CME feature licenses, and phone licenses purchased because of the fewer number of users in the office. A smaller office size may also dictate the selection of analog connectivity to the PSTN rather than digital connectivity.

Retail Business

The retail sector is a highly competitive market. Retailers face intense cost and competitive pressures from the many players that exist in this space. In addition, the number of large discount stores and Internet-based companies in this market has grown, leading to price pressures that benefit customers but not retailers. Because of this, margins tend to be low. Retailers look for every opportunity to cut costs and increase per-store productivity and operations. One of the ways to decrease operational costs and increase productivity is with a streamlined IP telephony network.

Current Retail Networks

Retailers with a large number of stores and a QoS-enabled WAN are likely to deploy a centralized Cisco CallManager system for their needs. Many smaller retailers may prefer Cisco IPC Express for their call processing needs, particularly those with a limited need for store-to-store calls and infrequent communications with headquarters. WAN connections in the retail industry tend to be very low-bandwidth (56 kbps or less), lack QoS, or, in some cases, traverse the Internet via a VPN.

PSTN connectivity for retail stores depends on the size of the stores. FXO is common for smaller stores, and T1 is common for larger stores. Retailers' communication needs revolve around phones, because few businesses use PCs or laptops for the average employee in the store. Phone communication scenarios involve incoming calls from customers who are unable to stop by the store and who want to inquire about a particular item or about store hours or who want to speak to a particular person or department.

In a typical call scenario, an employee answers most incoming calls, because the human touch is important for business goals. The employee answers the calls, puts the caller on hold, uses a paging system to contact the right department or person, and then transfers the call to the proper department or employee who can help the customer.

Because a retailer's call processing system is the main source of communication with customers, the type of features supported dictates which communication system is purchased. A number of features are advantageous. These include paging, speed dial, call park and picking up a call, hookflash transfer to free up PSTN FXO trunks, multiple-line appearances, and support for wireless phones.

A store typically would use shared-line appearances on its phones, meaning that when a certain department receives a call, all the phones in that department ring. This allows any sales representative not assisting customers to pick up the call.

Voice mail systems typically are not used during regular business hours, because retailers strive to answer calls as they come in and to process the customer query immediately. Voice mail may be useful to certain store employees, such as managers and supervisors, to aid in communication with headquarters (for example, a broadcast message about new sales goals or special incentives).

Certain stores have a full-time manager who supervises the employees, walks around the store, and uses the phone

Other Cisco IP Telephony Solutions for the Enterprise Branch and Small and Medium Offices

The Cisco IP telephony call processing options for the enterprise branch and small or medium office include Cisco CallManager with Cisco SRST and Cisco IPC Express, which you learned about earlier in this chapter.

Cisco CallManager is made up of call processing software running on a Cisco Media Convergence Server (MCS). Cisco SRST is a backup call processing service providing local telephony features in the event of a connectivity failure between a centralized Cisco CallManager and the remote IP phone in a branch office.

This section provides an overview of the Cisco CallManager and SRST solutions. Although this is not the focus of this book, the goal is to provide enough information so that you can decide whether Cisco IPC Express or Cisco CallManager and SRST is the right technology choice for your office or customer.

Introducing Cisco CallManager and SRST

Each Cisco CallManager call processing server can support up to 7500 IP phones. Several of these servers can be clustered to support large, scalable networks of up to one million users in a system with more than 100 sites.

Although Cisco CallManager is best suited to large enterprise networks with centralized call processing, a single server can also be deployed in a medium or large office. This solution is less cost-effective than Cisco IPC Express for smaller numbers of users; however, it may still be appropriate for the office that requires any of the more sophisticated call processing features available on Cisco CallManager. Some features supported with Cisco CallManager include unified messaging, multimedia conferencing, collaborative contact centers, and interactive multimedia response systems.

Cisco SRST is used in a centralized Cisco CallManager call processing architecture. Cisco SRST is Cisco IOS-based call processing software that stays dormant in the local branch office telephony router until the WAN experiences an IP connectivity failure.

When the branch office loses connectivity to the central Cisco CallManager because of a WAN or other IP connectivity failure, SRST kicks in to provide emergency backup call processing services for the phones in the branch office. SRST delivers a long list of basic call processing features for the duration of the failure, ensuring that the phones stay up and operational.

When IP connectivity is restored in the network, the system automatically reverts call processing functions to the centralized Cisco CallManager.

A number of books describe Cisco CallManager and SRST network operation in more detail, including the following:

- Troubleshooting Cisco IP Telephony by Paul Giralt, Addis Hallmark, and Anne Smith. Cisco Press. 2002.
- Cisco CallManager Fundamentals: A Cisco AVVID Solution by John Alexander, Chris Pearce, Anne Smith, and Delon Whetten. Cisco Press. 2001.

Deciding Between Cisco IPC Express and Cisco Call Manager

You have to consider many factors when deciding between Cisco IPC Express and Cisco CallManager for the IP telephony application for your office or network. First consider the WAN environment. If the WAN has not been upgraded to deliver QoS protection for voice, or if there is limited WAN bandwidth, or if a WAN does not exist, a locally delivered call processing service is best for remote offices. In this scenario, Cisco IPC Express delivers a very cost-effective solution.

Summary

In this chapter, you learned about Cisco IPC Express, its cost savings and productivity benefits, the components that make up the system, and some of the features you may use in your business. You also learned how to select between Cisco IPC Express and other Cisco IP telephony options. Examples were provided of how Cisco IPC Express applies to businesses in different industries.

This chapter focused on the business or office level, the pieces of a Cisco IPC Express system, and the features benefiting end users. In the next chapter, the focus turns to the network level. You will learn about the different types of networks on which Cisco IPC Express can be deployed.

Subsequent chapters provide in-depth details on the Cisco IPC Express system architecture, deployment models, and features, and how to install, configure, and manage it for your business.

Chapter 2. Building a Cisco IPC Express Network

This chapter covers the following topics:

- IP Telephony network deployment models
- Cisco IP Telephony system trade-offs
- Cisco IP Communications (IPC) Express deployment models, including the standalone office, the multisite network, and the service provider managed services models

Chapter 1, "Introducing Cisco IPC Express," provided an overview of why Cisco IP Communications (IPC) Express might be an excellent voice and data communications system for your business and how its features may apply to your needs. It also gave brief information about the router platforms, licenses, and components that comprise Cisco IPC Express. This chapter covers what your network might look like or, as it is more formally called, what deployment models you may consider when building a network that contains one or more sites with Cisco IPC Express.

If you own a small business that operates from a single site, your network is equally simple, so only a small portion of the information here applies to you. On the other hand, perhaps yours is a larger business with multiple sites, or you own (or work for) a firm that has an enterprise backbone network and you are considering Cisco IPC Express for some or all of the branch sites. If so, you will find the network deployment information in this chapter handy when connecting Cisco IPC Express to your existing network. Last, if you are a service provider (SP) considering Cisco IPC Express as a hosted or customer premises equipment (CPE) offering to your customers, your network has even more considerations.

IP Telephony Network Deployment Overview

Before covering Cisco IPC Express-specific deployment details, you should understand a little more about general IP telephony networks and some of the trade-offs when selecting what type of network fits your business best. This fundamental understanding is needed because Cisco IPC Express networks follow the same general architectures and trade-offs. They also represent a subset of the larger canvas of IP telephony options you may choose from for your business or branch office. Another reason to understand general IP telephony networks is because you can mix and match systems of various types in the same network. If you are already familiar with IP telephony network architectures in general, you can safely skip this introductory section, and proceed directly to the Cisco IPC Express deployment models covered in the next section.

The brain of a telephony network is the call control (or call processing) component. This component of the network generally can be located anywhere in the network in one or multiple places. It provides call features to phones such as dial tone, digit interpretation to implement a dialing plan, and setting up and tearing down voice calls (or, more technically, a speech path or media stream) from the calling user to the called user. The call control component also manages supplementary features such as call hold, transfer, conference, music on hold, call waiting tone, and the myriad voice call features you are already familiar with.

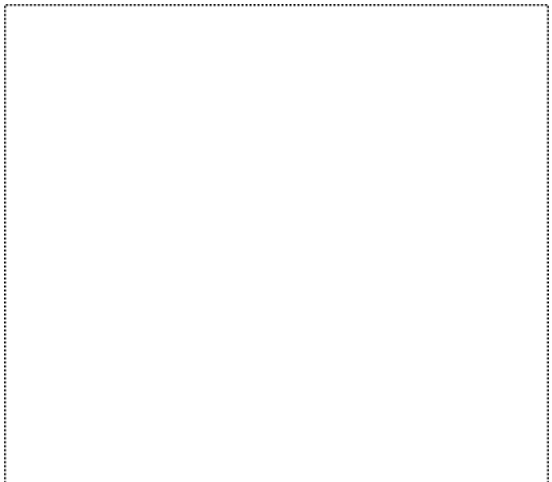
When considering where and how call processing is provided to users and phones, IP telephony networks can be classified broadly into the following types. Each of these is discussed in the subsequent sections:

- Single-site or standalone network
- Centralized network
- Distributed network
- Hybrid network

Single-Site or Standalone Network

Single-site or standalone networks are businesses or networks where all the employees are located at a single site, as shown in Figure 2-1. By definition, these types of offices or networks are predominantly small, often fewer than 30 people, and occasionally perhaps up to 100 or 150 people. Any organization larger than this most likely has multiple geographic sites.

Figure 2-1. Single-Site Call Processing Network



Cisco IP Telephony System Trade-Offs

For the small standalone office, Cisco IPC Express is an excellent system. For larger networks of multiple sites, the decision to centralize or distribute call processingor, more specifically, which sites should get call processing servers and which should access services from a nearby larger hub locationis key to the fundamental design of an IP telephony network. It is also one of the important factors dictating whether Cisco IPC Express is the best fit for each individual site.

Generally, two Cisco IP telephony products should be considered for the call processing component in a standalone or multisite network or a section of that network:

- Cisco CallManager A separate Intel-based server (or multiple servers in a redundant configuration) available in a range of capacities from 1000 to 7500 IP phones. You can cluster these servers to form server farms with increased redundancy and availability to provide IP telephony to large campuses and to act as a hub for up to 500 remote sites. Although individual return on investment (ROI) varies, Cisco CallManager generally is not cost-effective at sites of fewer than approximately 500 phones. You can deploy Cisco CallManager in a centralized, distributed, or hybrid model.
- Cisco IPC Express A Cisco IOS-integrated call processing component that is coresident in the router at the site. Cisco IPC Express is cost-effective at the low end and can provide IP telephony services at sites requiring up to 240 phones. For a multisite network, Cisco IPC Express inherently represents a distributed call processing model, because the call control is built into the router at each site.

Cisco also offers other call processing components, such as the Cisco BTS 10200 Softswitch (on Cisco.com, search for "BTS 10200") and the Cisco PGW 2200 Softswitch (on Cisco.com, search for "PGW 2200"). However, these are used primarily in SP networks for residential or large-scale voice services, such as a long-distance carrier service. They are not deployed as the call processing component in a standalone business or small or medium enterprise, which is the focus of this book. Therefore, these products are not covered here in any further detail.

Considering the Cisco CallManager and Cisco IPC Express products, the network design decision at hand is not only centralized versus distributed, but also which product is the best choice for which site. The following sections explore the key trade-offs to help you determine which design might best fit your network.

Cisco Call Manager Networks

Many large enterprises comprising hundreds or thousands of sites find a centralized Cisco CallManager network the best option. Here the term centralized is used in a looser sense than in the preceding sections, meaning that most phones and sites draw their call processing services from another, larger site. But inevitably, multiple Cisco CallManager servers are distributed among several of the largest, central locations, and the network is, strictly speaking, a hybrid network. The multiple servers are required for both scalability and reliability. Thus, the core of the network is a distributed model, whereas the smaller sites follow a centralized model.

In these types of networks, voice mail is often also centralized, provided by large-scale servers with thousands or tens of thousands of users per server or server farm. Many other applications, such as e-mail and business applications (for example, product ordering systems and web servers), might also be located at the same site and concentrated in the same data center(s). Therefore, they are a centralized resource to the majority of the remote sites in the network.

Providing both call processing and voice mail services in a centralized manner requires the following network and business attributes:

- Sufficient WAN bandwidth for voice calls from the remote sites to the central site where the call processing or voice mail servers are hosted

Understanding Cisco IPC Express Deployment Models

This section explores Cisco IPC Express network deployments in greater detail. This assumes that one of the Cisco IPC Express models described in the preceding sectiona pure Cisco IPC Express network with one or more interconnected sites, or a hybrid Cisco CallManager and Cisco IPC Express networkis a good fit for your organization.

Three general deployment models are discussed here:

- Standalone office A single-site business with typically fewer than 100 employees.
- Multisite business A branch or remote office interconnected to other sites in the same network.
- Service provider (SP) managed services Can be either of the previous two categories. Instead of the company buying and managing the equipment for its own services, it pays a recurring charge to a local SP. This SP owns the equipment and hosts the services, typically both voice and data services.

Standalone Office

The standalone office model fits the vast number of small, single-site companies throughout the world that have fewer than 100 employees. Here are some examples:

- A dentist's office
- A small health clinic
- A professional services office for architects, lawyers, or interior decorators
- A small charity
- A florist shop with three locations in the neighborhood

These types of businesses have no IT organization, and their handful of employees are focused on conducting the company's core business. They are dentists, dental assistants, plumbers, lawyers, florists, or car mechanics, not IT professionals. Data and voice services supporting these types of businesses are either hosted by a local SP or installed and maintained on the premises by a local value-added reseller (VAR) or systems integrator (SI).

IP telephony can be as advantageous to this type of small business as it can to a large enterprise with considerable IT expertise. Voice services to small companies such as these traditionally have been provided either by centrex services or by a key system installed by a VAR or SI. Data service was provided by the local Internet service provider (ISP), which placed some type of CPE onsite and managed it from a central point.

For any site except the very smallest of perhaps only one or two employees, the CPE equipment would include a router and basic security services, such as a firewall.

The next several sections explore different aspects of the standalone office network, including the network architecture, applications, management, and security services deployed.

Summary

In this chapter, you learned how and where to deploy Cisco IPC Express technology to its best effect for your business. You gained insight into how Cisco IPC Express fits into different types of networks and the business trade-offs for choosing Cisco IPC Express or other Cisco IP telephony options. This chapter highlighted the PSTN connectivity, WAN connectivity, applications, management, and security considerations for different types and sizes of networks. It also explored how technology and network architecture underscore your business management philosophy and decisions about network connectivity and bandwidth provisioning. Furthermore, you saw how your network might have to evolve to support IP telephony solutions such as Cisco IPC Express.

The next chapter goes into considerable detail about the features, operation, and multisite interconnection options of Cisco IPC Express, including Cisco CallManager Express (CME) and Cisco Unity Express (UE).

Chapter 3. Cisco IPC Express Architecture Overview

This chapter covers the following topics:

- Cisco IP Communications (IPC) Express system architecture
- Cisco CallManager Express (CME) architecture
- Cisco Unity Express (UE) applications architecture

An IP Communications (IPC) Express system is a complete, integrated communications system for a small- or medium-sized standalone or networked office site. Fundamental to its architecture is the Cisco IOS router on which all the data and voice services offered by the system are based.

This chapter first explores the internal architecture of the Cisco IOS router on which Cisco CallManager Express (CME) is based. Then it takes a closer look at the internal architecture of the Cisco Unity Express (UE) application and its interaction with Cisco CME and the router. The packaged combination of Cisco CME and Cisco UE is called Cisco IP Communications (IPC) Express.

If you are already familiar with IP-based networks, you might approach your Cisco IPC Express system primarily as a Cisco IOS router with telephony capability. On the other hand, if you are looking at the Cisco IPC Express system as an integrated, single-box office communications system, the router technology included in the platform might be a secondary consideration for you. As covered in Chapters 1 and 2, Cisco IPC Express is supported on a range of low-end and mid-range router platforms. You can choose from among these to provide the appropriate level of connectivity and processing power for the services your office requires.

Cisco IPC Express is composed of two main components:

- The call processing engine, called Cisco CME (including communications endpoints, such as the IP phones)
- Optional applications such as automated attendant (AA), voice mail (offered by Cisco UE), and IP phone-based XML and Telephony Application Programming Interface (TAPI) applications provided by external application servers

This chapter takes an architectural overview of these components and how they interact.

Cisco IPC Express System Architecture

Figure 3-1 shows the overall software architecture of a Cisco IPC Express system. Cisco CME and its internal components are on the left, and Cisco UE and its components are on the right. The Cisco CME software (the call processing software) runs as part of Cisco IOS on the host router platform. Cisco UE runs in a Linux-based embedded environment on a hardware module that slots into the host router.

Figure 3-1. Cisco IPC Express System Architecture

[View full size image]

On a physical level, Cisco CME and Cisco UE communicate via an IP (Ethernet) interface across the backplane of the router chassis. As shown in Figure 3-1, there are several logical connections between the two components:

- A graphical user interface (GUI) browser interface that communicates with the router's HTTP server as well as HTTP server software on the Cisco UE module
- System management communication such as configuration lookup during the Cisco UE Initialization Wizard, and configuration synchronization with the router during the life of the system
- Call control communication using a Session Initiation Protocol (SIP) interface
- Underlying communications connectivity via TCP and Ethernet

Cisco CME Architecture

Historically, the evolution of voice support on Cisco routers first included Public Switched Telephone Network (PSTN) and Private Branch Exchange (PBX) connectivity into voice-over-packet networks, the traditional toll bypass voice over IP (VoIP) network architectures. This connectivity to legacy systems (the PSTN and PBXs) is often called PSTN gateway or voice gateway functionality. Support for IP phones followed next in the evolution with the development of Survivable Remote Site Telephony (SRST). When all these technologies were in place, a comprehensive standalone telephony solution such as Cisco CME was a natural next step. All the Cisco CME phone features are built on top of these two foundation elements: the router engine and the VoIP infrastructure first developed to provide PSTN and PBX gateway connectivity on a router. This is an important perspective to embrace in understanding Cisco CME and its architecture.

This inheritance has some advantages and also some disadvantages. The primary advantages include the fact that Cisco CME leverages the extensive base of Cisco IOS IP routing and voice technologies. This lets Cisco CME interoperate with a large installed base of IP network types and different voice protocols and technologies. It also means that new developments in the Cisco IOS IP and voice technology area tend to get bundled automatically with each new release of the Cisco CME software. This significantly reduces the risk of rapid marketplace obsolescence inherent in any new technology. Considering the ongoing competition in the VoIP marketplace between the International Telecommunication Union (ITU)-sponsored H.323 and Internet Engineering Task Force (IETF)-sponsored SIP technologies, this has particular importance for anyone concerned about picking the "wrong" technology. Because Cisco CME is based on Cisco IOS voice technology, it includes both H.323 and SIP support. It will continue to evolve as these technologies unfold in the industry.

The disadvantage of this integrated approach is that the Cisco CME platform was not originally designed to be a phone system. It was designed as a router with the ability to handle voice traffic. This means considerations and compromises are reflected in how Cisco CME is designed, configured, and managed and in how it operates. How you view these trade-offs may depend in part on whether you are looking for a phone system as a standalone technology purchasing decision or as part of a larger communications infrastructure and longer-term investment.

If you are approaching Cisco CME from the point of view of an integrated and converged voice and data communications infrastructure, you are more likely to see the extra complexity and capabilities that come with the Cisco IOS foundation as a good thing and as necessary to meet your business networking needs. The complexity that's inherent in Cisco IOS isn't arbitrary. It has evolved over many years of dealing with the real-world network intricacies that exist when you try to interconnect multiple pieces of equipment, and evolve your network in several phases over an extended period of time.

Cisco CME Software Architecture

The Cisco CME phone features in Cisco IOS are built on top of the Cisco IOS voice infrastructure foundation. This gives Cisco CME access to both H.323 and SIP interfaces. It also lets Cisco CME be protocol-agnostic regarding H.323 versus SIP for VoIP calls across intersite links. The protocol used to control local IP phones is the lightweight Cisco Skinny Client Control Protocol (SCCP), often simply called "Skinny." This protocol gives Cisco CME tight control over the IP phones and lets it offer a rich set of phone features from a call control, phone user interface, and provisioning point of view.

SCCP is a Cisco-specific interface that is used only internally between the Cisco CME router and the IP phones it controls. The external interface that Cisco CME exposes to phone connections across its public IP link to other sites uses the standards-based H.323 or SIP protocols. Figure 3-2 shows the VoIP protocols used by a Cisco CME system.

Figure 3-2. Protocols Used by a Cisco CME System

Cisco UE Applications Architecture

Cisco UE was designed as an applications environment from the ground up. It is not as heavily based on the Cisco IOS router architecture as the Cisco CME software. The Cisco UE hardware module is a Linux-based environment with local storage and CPU capacity that communicates with the host router via a logical IP interface. From a communications and system architecture point of view, the interface between the Cisco UE hardware and the host router looks like an Ethernet interface to both sides. This is true even though there is no physical cabling, and the communication runs across the backplane of the router chassis.

Although Linux is the operating system (OS) underlying the Cisco UE software, it is an entirely embedded system. It is invisible and inaccessible to the system's external interfaces (CLI, GUI, IP). Likewise, Cisco UE's database and Lightweight Directory Access Protocol (LDAP) directory components are purely internal to the architecture; they have no external access or interfaces.

Cisco UE can be deployed with Cisco CME or Cisco CallManager as the call control agent. Only the Cisco CME deployment architecture is within the scope of this book. Go to http://www.cisco.com/go/cue for more information on how to deploy Cisco UE in Cisco CallManager/SRST networks.

Cisco UE Hardware Architecture

The Cisco UE application is based on an Intel Pentium-based hardware module that fits into the router chassis. It is available in two form factors:

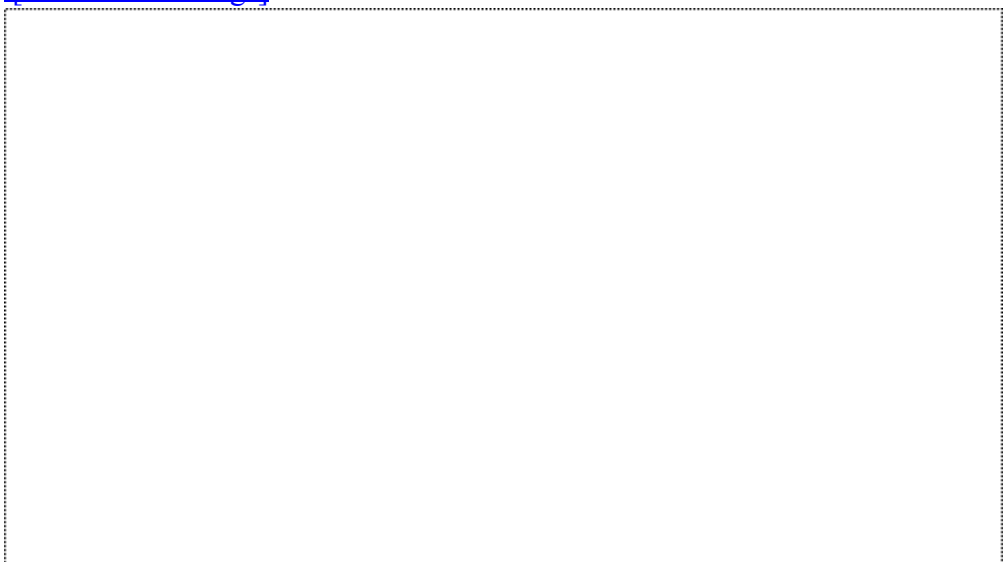
- Network module (NM-CUE)
- Advanced Integration Module (AIM-CUE)

The NM form factor contains a hard disk for persistent storage, whereas the AIM form factor uses an industrial-strength compact Flash card.

The Cisco UE hardware module (irrespective of form factor) has a back-to-back Fast Ethernet connection to the host router. The host router sees an Ethernet interface through its Peripheral Component Interconnect (PCI) backplane that is internally hard-wired to a second Ethernet interface. The second interface (called the service-engine interface) is controlled by the Cisco UE module, as shown in Figure 3-8.

Figure 3-8. Router IP Interface to Cisco UE Hardware Module

[View full size image]



Summary

This chapter provided an architectural overview of the two main Cisco IPC Express components: Cisco CME, the router-based call processing engine, and Cisco UE, the AA and voice mail applications engine.

The Cisco CME architecture was discussed within the larger scope of Cisco IOS-based voice services. This chapter covered how Cisco CME features and services fit within, and reuse, fundamental Cisco IOS voice infrastructure, such as dial peers and PSTN trunk interfaces. It also discussed how the Cisco CME software handles the different VoIP protocols, including SCCP, H.323, and SIP. This included considerations such as firewalls and NAT.

Cisco UE hardware is a module that fits within the router. You learned how this hardware communicates with the router hardware and software. The Cisco UE applications environment is based on Linux. The software components, structure, and interactions between subsystems also were explained.

All this information provides a solid understanding for the fundamental components and functions of a Cisco IPC Express system. The chapters in Part II of this book describe in much more detail exactly what the Cisco IPC Express features are, how they can work in your business, and how you can configure and manage the system to your best advantage. Part III draws on this architectural information, because a fundamental understanding of the system components is necessary to troubleshoot the system.

Recommended Reading

For additional information on Cisco IOS voice, consider the following books:

- Troubleshooting Cisco IP Telephony by Paul Giralt, Addis Hallmark, and Anne Smith. ISBN 1587050757.
- Voice over IP Fundamentals by Jonathan Davidson, James Peters, and Brian Gracely. ISBN 1578701686.
- Integrating Voice and Data Networks by Scott Keagy. ISBN 1578701961.

Part II: Feature Operation and Applications

Chapter 4: Cisco IP Phone Options

Chapter 5: Cisco CME Call Processing Features

Chapter 6: Cisco CME PSTN Connectivity Options

Chapter 7: Connecting Multiple Cisco CMEs with VoIP

Chapter 8: Integrating Cisco CME with Cisco CallManager

Chapter 9: Cisco IPC Express Automated Attendant Options

Chapter 10: Cisco IPC Express Integrated Voice Mail

Chapter 11: Cisco CME External Voice Mail Options

Chapter 12: Additional External Applications with Cisco CME

Chapter 4. Cisco IP Phone Options

This chapter covers the following topics:

- Features and capabilities of the following Cisco IP Phones: 7970G, 7940G, 7960G, 7914 Expansion Module, 7910G, 7905G, 7912G, 7902G, 7935 and 7936 Conference Stations, and 7920 wireless phone
- Cisco Analog Telephony Adapter (ATA)
- Cisco IP Communicator softphone
- Firmware files for the IP phones
- Resetting and restarting the IP phones
- Erasing the IP phone configuration
- Accessing call parameters for an active call

Cisco CallManager Express (CME) supports a range of phones that meet various customer needs, from the simple low-end Cisco 7902G IP Phone to the advanced Cisco 7970G color touch-screen phone, the 7936 Conference Station, and the Analog Telephony Adaptor (ATA) that lets you connect analog phones to Cisco CME. This chapter discusses the various phones supported by Cisco CME to give you a better understanding of how these devices operate and which features are supported. After reviewing the main hardware features of each phone, you learn the configuration steps necessary for each phone type. This chapter also covers some phone-specific information such as firmware loads, resetting the phone, erasing the phone configuration, and accessing call parameters from the phone.

The Cisco 7940G and 7960G IP Phones

The Cisco 7940G and 7960G, shown in Figure 4-1, were the first phones available with Cisco CME. These phones support all the features currently supported with Cisco CME 3.2.

Figure 4-1. Cisco 7940G and 7960G IP Phones

Note

Although this chapter refers primarily to the current G-series phones (such as the 7960G and 7940G), Cisco CME also supports the previous generation of phones, such as the Cisco 7960 and 7940. The only difference between the G and non-G phones is in how their buttons are labeled: with English words (the non-G phones) or with language-independent icons (the G series).

Hardware

The Cisco 7960G and 7940G IP Phones look identical, except for the number of buttons on each phone. The Cisco 7940G has two line buttons, whereas the Cisco 7960G has six line buttons and can have an additional 28 lines with the addition of up to two Cisco 7914 expansion modules. The phones require 48V DC power for operation.

The Cisco 7960G and 7940G IP Phones include a 10/100 Ethernet switch port that allows a PC to be connected via the phone's Ethernet connection. The two Ethernet ports on the phone are marked as 10/100SW and 10/100PC. The 10/100SW should be connected to the Ethernet switch port, and the 10/100PC port can be connected to a UNIX workstation or PC, as shown in Figure 4-2.

Figure 4-2. Connecting a PC to an IP Phone

Both phone ports are capable of autonegotiation. 48V power is required for the phone to operate. This can be provided by an external power supply unit or via a LAN switch with inline power capability. The inline power LAN switch might be an external switch or an integrated Layer 2 LAN switching module resident in the network module or high-speed WAN interface card (HWIC) slot of the Cisco CME router. Only the 10/100SW port can receive inline

The Cisco 7914 Expansion Module

The Cisco 7914 Expansion Module is an add-on device that attaches to the Cisco 7960G IP Phone to increase the number of line buttons. Figure 4-3 shows a Cisco 7914 Expansion Module attached to a Cisco 7960G IP Phone.

Figure 4-3. Cisco 7960G with a 7914 Add-on Module

Hardware

The Cisco 7914 expansion module has 14 buttons and a single-line LCD screen for each button. Each module adds 14 lines to the Cisco 7960G host phone that it is attached to. You can add a maximum of two expansion modules to a Cisco 7960G IP Phone and, thus, increase the phone's total lines to 34.

A separate 48V DC power supply is required for the Cisco 7914 module. Inline power from an Ethernet switch cannot provide power to a Cisco 7914 module. An RS-232 cable provides the communication link between the Cisco 7914 module and the Cisco 7960G IP Phone. The buttons are lit amber, red, or green, depending on the state of the line associated with the button.

Configuring the Cisco 7914 Expansion Module

You can configure the 14 lines on the Cisco 7914 module as individual lines, speed dials, intercom, or monitored lines. The Cisco 7914 module is not configured as a separate device under Cisco CME; hence, it doesn't require the network parameter configuration required for other IP phones.

The Cisco 7914 module does, however, require firmware for its operation. The firmware for the module depends on the version of Cisco CME you are using. It should be configured under the telephony-service configuration prompt using the load command. The buttons are considered a continuation of the buttons on the Cisco 7960G IP Phone to which it is attached. Hence, the first button on a Cisco 7914 is configured as the seventh button on the Cisco 7960G IP Phone, and the 14th button of the Cisco 7914 is configured as the 20th button. Example 4.2 shows a typical configuration for a Cisco 7960G IP Phone with a Cisco 7914 module attached.

Example 4.2. Cisco 7914 Configuration

```
Router#show running-config
telephony-service
max-dn 10
max-ephone 10
ip source address 10.0.0.1 port 2000
load 7940-60 P00303020214
load 7914 OSX001124351
create-cnf file
ephone-dn 1
  number 1001
ephone-dn 2
  number 1002
..
..
```

The Cisco 7910G IP Phone

The Cisco 7910G IP Phone, shown in Figure 4-4, is a low-end phone that is part of the same family of phones as the Cisco 7960G and 7940G. The primary difference is that the Cisco 7910G phone has only a two-line display and no softkeys. The Cisco 7910G supports English only and does not support XML capabilities.

Figure 4-4. Cisco 7910G IP Phone

Hardware

The Cisco 7910G IP Phone comes in two flavors: the Cisco 7910G and the Cisco 7910G+SW models. The Cisco 7910G has a standard 10 Mbps Ethernet interface, and the Cisco 7910G+SW is equipped with two 10/100 Ethernet interfaces. The ports are marked as 10/100SW and 10/100PC, respectively. The functionality of the ports is similar to the Cisco 7960G IP Phone, where one port can be connected to a LAN switch and the other port can be connected to a PC.

The Cisco 7910G supports only a single line (extension) and has a two-line, text-only LCD screen. The phone does not have a microphone, although it does have a "listen-only" speakerphone. If you put a call on "listen-only" speakerphone, the far-end party can't hear you. The phone has fixed function buttons for two speed dials, redial, hold, conference, transfer, accessing voice mail, and configuring call-forward from the phone.

Configuring the Cisco 7910G IP Phone

Configuring network parameters for a Cisco 7910G is similar to doing so for a Cisco 7960G IP Phone. The phone's default behavior is to use DHCP to obtain the network parameters. You can also manually configure these parameters from the phone by unlocking the phone and disabling DHCP services first.

Pressing the Settings button takes you through a menu with various options. This menu is comparable to the Settings menu for a Cisco 7960G. Because of the limited display and softkeys available, some of the functions on this menu work differently on a Cisco 7910G IP Phone than on a Cisco 7960G.

To unlock the phone, press the key combination **# before pressing the settings button. You can see the lock's status in the top-right corner of the LCD screen after pressing the settings button.

The menu gives you the following options, with a number assigned to each option:

- Handset Vol
- Speaker Vol
-

The Cisco 7905G and 7912G IP Phones

The Cisco 7905G and 7912G are entry-level IP phones. Figure 4-5 shows the Cisco 7905G IP Phone. The physical appearance of these two phones is very similar, except for the additional Ethernet port available on the Cisco 7912G.

Figure 4-5. Cisco 7905G IP Phone

Hardware

The Cisco 7905G has a single 10-Mb Ethernet interface. The device requires 48V DC power to operate, which can be provided from an inline Cisco LAN switch or an external power supply unit. The Cisco 7905G has a pixel-based LCD screen.

Note

The Cisco 7905G IP Phone is the only model where Cisco CME supports only the G-series model and not the ordinary Cisco 7905 IP Phone. The Cisco 7905 is an H.323 phone, and Cisco CME supports only SCCP-based phones (such as the Cisco 7905G).

The Cisco 7912G has two 10/100 Mbps Ethernet ports and functions similar to the Cisco 7960G phone. The power supply can be provided from an inline power LAN switch or an external power supply.

Both the Cisco 7905G and 7912G phones support a single-line button and have an additional four buttons for hold, transfer, conference, and dialing voice mail. The phone is also equipped with a "listen-only" speakerphone, mute button, volume control keys, and a rocker button to scroll through the LCD screen. The function of the softkeys changes depending on the phone's state. The softkeys can be programmed for various states of the phone, such as the Cisco 7960G. The hold function button is not a part of the softkey set. A dedicated button for this function is located on the left side of the rocker button; it has a hexagon symbol resembling a stop sign. The button on the right side of the rocker button takes you to the Settings menu, similar to the phones discussed previously.

Configuring the Cisco 7905G and 7912G IP Phones

The Cisco 7905G and 7912G IP Phones obtain their IP address and other network information by default from DHCP. You can also configure these parameters from the phone after disabling DHCP. The phone is equipped with a set of menus similar to the Cisco 7960G IP Phone to view and configure some of the phone and network parameters. The key combination to unlock and reset the phone is the same as for the Cisco 7960G IP Phone.

The phone load and phone type should be configured under the telephony-services and ephone configuration modes, respectively, for a successful registration of the phone. The phone can be configured with two single-line directory numbers (DNs) or a single dual-line DN.

The Cisco 7970G IP Phone

The general look and feel of the Cisco 7970G IP Phone, shown in Figure 4-6, is similar to that of a Cisco 7960G, with added buttons, features, and a color display.

Figure 4-6. Cisco 7970G IP Phone

Hardware

The Cisco 7970G IP Phone has an LCD color display that is bigger than the Cisco 7960G display and has touch-screen capability. The phone has eight line buttons and five softkeys. The line buttons are light emitting diode (LED) lit with different colors indicating different phone states (for example, ringing or hold).

The phone's network capabilities are similar to that of a Cisco 7960G IP Phone. The phone also has a sleep/wake button for the LCD screen. The phone needs a 48V DC power supply to operate; it can be provided from a LAN switch capable of inline power or from an external power supply. If the phone is powered from an inline power LAN switch, the LCD screen does not operate with full brightness (this requires an external power supply). The Cisco 7970G IP Phone supports 802.3af inline power.

Configuring the Cisco 7970G IP Phone

The Cisco 7970G is supported on Cisco CME 3.2.1 and later. Configuring a Cisco 7970G is the same as with a Cisco 7960G. The phone requires certain firmware depending on the version of Cisco CME you are using. This firmware version is configured in telephony-service configuration mode.

You also can configure the phone manually from the Settings menu. You can configure the system message and softkeys for various phone states from the Cisco CME router console.

The Cisco 7902G IP Phone

The Cisco 7902G, shown in Figure 4-7, is the lowest-end IP phone in the portfolio supported by Cisco CME. The phone's physical appearance is similar to a Cisco 7905G IP Phone but without an LCD screen.

Figure 4-7. Cisco 7902G IP Phone

Hardware

The default configuration of network parameters for a Cisco 7902G IP Phone is via the DHCP mechanism. Because of the lack of display, manual configuration is carried out via a built-in English-only interactive voice response (IVR) menu on the phone. The Cisco 7902G has a single 10-Mb Ethernet interface and requires a 5V DC external power supply to operate.

The four fixed function buttons provide redial, transfer, conference, and voice mail functionality. You can configure up to ten speed dials for a Cisco 7902G. To place a call using speed-dial, press and hold (for about two seconds) the number on your keypad (0 to 9) assigned as a speed-dial number.

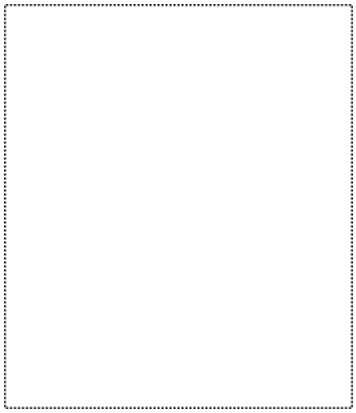
Configuring the Cisco 7902G IP Phone

The Cisco 7902G IP Phone has hold and function keys. Pressing the hold button puts an active call on hold. The same action retrieves a call if one is on hold. Pressing the function button when the phone is off-hook activates the IVR menu that allows you to manually configure the phone's various network and service parameters.

The Cisco 7935 and 7936 IP Conference Station

The Cisco 7935 and 7936 IP Conference Stations are IP-based, hands-free conference room stations. Figure 4-8 shows the Cisco 7935 IP Conference Station.

Figure 4-8. Cisco 7935 IP Conference Station



Hardware

The Cisco 7935 and 7936 conference stations provide a speakerphone for a large conference room. The general look and feel of the Cisco 7935 and 7936 are the same. Both have 10/100 Mbps Ethernet interfaces to connect to a LAN switch and operate from a 19V DC external power supply. The Cisco 7935 and 7936 do not support inline power.

The conference station has standard dial pad, off-hook, volume control, and mute buttons. Additional buttons are available on the phone to configure and navigate through the network and phone parameters.

The conference station does not have a handheld receiver. Hence, the phone's off-hook button should be used to make a call. The Cisco 7935 can cover a 10 ft. by 12 ft. (3 m by 3.6 m) conference room, and the Cisco 7936 can support a room as big as 20 ft. by 30 ft. (6 m by 9.1 m). The Cisco 7936 has two external microphones that can be connected to the main unit.

Configuring the Cisco 7935 and 7936 Conference Stations

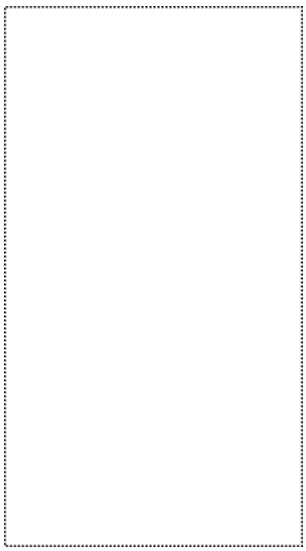
Configuring a Cisco 7935 or 7936 conference station is similar to doing so with the IP phones mentioned in the previous sections. You can manually configure the phone's network parameters from the phone in a similar fashion. The Cisco 7935 and 7936 should be configured with their respective device types when configured under Cisco CME.

The menu button on the conference station provides access to network and phone-specific parameters. This menu is slightly different from the one on the phones discussed in the previous sections. You can access the network parameters by selecting the option admin setup. This menu is password-protected; it comes with a factory default password of 12345. As soon as you have access to this menu, you can change or edit the conference station's network parameters.

The Cisco 7920 Wireless IP Phone

The Cisco 7920 Wireless IP Phone, shown in Figure 4-9, is an 802.11b wireless communication device supported by Cisco CME.

Figure 4-9. Cisco 7920 Wireless IP Phone



Hardware

The Cisco 7920 Wireless IP Phone needs a Cisco Aironet 802.11b access point to work with Cisco CME. The phone uses a Lithium-ion battery for power; it can be charged using an external AC adaptor.

This phone has the following buttons:

- A standard dial pad button
- A left and right key that activates the softkey above it
- An answer/send key to answer a call or send the number
- Arrow keys to navigate through the phone menu
- A power/end key to turn off the phone, silence a ringing call, or end a connected call

You can configure up to six lines on the Cisco 7920 Wireless IP Phone.

Configuring the Cisco 7920 Wireless IP Phone

The Cisco 7920 Wireless IP Phone is supported by Cisco CME 3.0 and later. This phone is configured like any other wireless device. Go to http://www.cisco.com/go/wireless for more information on setting up your wireless network for voice traffic.

The Cisco CME configuration required for a Cisco 7920 Wireless IP Phone is no different from any of the other IP

The Cisco Analog Telephony Adaptor

The Cisco Analog Telephony Adaptor (ATA) is a small gateway device that provides an interface between Skinny Client Control Protocol (SCCP), used for IP phones, and a standard analog phone. Thus, the ATA lets you register an analog phone with Cisco CME as if it were an IP phone. The Cisco ATA device comes in two models: the Cisco ATA 186 and Cisco ATA 188 (shown in Figure 4-10). The models look similar from the front, but from the rear you can see the additional Ethernet port available on the ATA 188.

Figure 4-10. Cisco ATA 186 and ATA 188

Hardware

Both the ATA models connect one or two analog telephones. The hardware features for the ATA 186 include the following:

- Function button
- Two RJ-11 Foreign Exchange Station (FXS) ports
- RJ-45 10Base-T Ethernet uplink port
- Network activity (ACT) LED
- Power connector

One of the major differences between the two models is that the Cisco ATA 188 has two RJ-45 10/100-Mbps Ethernet ports, whereas the Cisco ATA 186 has only a single10 Mbps Ethernet port. One of the ports on the Cisco ATA 188 is an uplink port, and the other one is a data port.

The FXS interfaces can be used to connect two standard analog telephones or fax machines. The Cisco ATA is connected to the network via an Ethernet interface and can be configured via DHCP or manually. The Cisco ATA needs a 5V DC external power supply to operate. Inline Ethernet power does not apply to the Cisco ATA devices, because they do not support phones on Ethernet ports.

Note

Cisco CME 3.2 does not support a fax machine using SCCP. A fax machine on a Cisco ATA device must use H.323.

The Cisco IP Communicator Softphone

The Cisco IP Communicator is a software-based application that allows users to place and receive calls from their PCs. IP communication between your personal computer and Cisco CME should be in place for successful operation of Cisco IP Communicator. Also, the IP network connecting your PC should be designed to carry voice traffic. Figures 4-12 and 4-13 show the Cisco IP Communicator interface with the default skin and the alternate skin, respectively.

Figure 4-12. Cisco IP Communicator Default Skin

[View full size image]

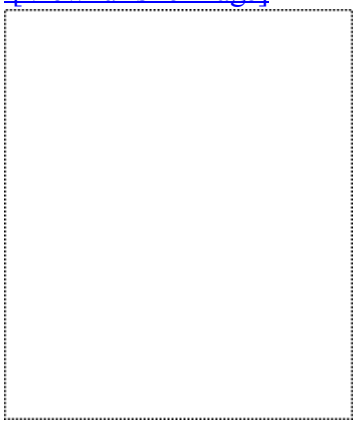


Figure 4-13. Cisco IP Communicator Alternate Skin

Hardware

The minimum PC hardware and software required for Cisco IP Communicator to function are as follows:

- Windows 2000 Professional with service pack 3.0 or later or Windows XP Professional with service pack 1.0 or later
- 450-MHz Pentium III or the equivalent

Firmware Files for IP Phones

Each type of IP phone requires firmware to register and function properly with Cisco CME. The version of the firmware depends on the type of phone and version of Cisco CME you are using. The firmware should be placed on a TFTP server accessible to the router and should be copied to the Cisco CME router. Also, the telephony-service configuration should have the firmware specified for each type of phone connected to the Cisco CME system.

The configuration shown in Example 4.2 lists the firmware for the Cisco 7940G, 7960G, and 7905G IP Phones. These files change depending on the version of Cisco CME you're using. Check Cisco.com for the correct version of these files for your system. The files in Example 4.2 are shown to illustrate where they are configured.

Example 4.3. Phone Firmware

```
Router#show running-config
telephony-service
max-dn 10
max-ephone 10
ip source address 10.0.0.1 port 2000
load 7940-60 P00303020214
load 7914 S00103020002.bin
load 7905 CP79050101SCCP030530B31.zup
```

All the phones come with factory default firmware installed. During the registration process, the phone compares its current firmware with the load configured under telephony services, as shown in Example 4.2. If the configured firmware and the current firmware match, the phone continues with the registration process. If the firmware is different from what is configured, the phone tries to download the new firmware from Cisco CME.

It is necessary to have the right version of phone firmware for the version of Cisco CME you are using to have all the features working as designed.

In certain cases, a firmware upgrade or downgrade is not a one-step process. Also, in certain cases, you cannot downgrade the phone firmware to a lower version. For example, you cannot downgrade the firmware on a Cisco 7960G IP Phone from a signed load to an unsigned load. This process is covered in much more detail in Chapter 16.

Resetting and Restarting the Phones

You may have to reset or restart the phones when you make changes to the Cisco CME configuration. When a phone is reset, it goes through the entire bootup sequence. A phone reset is very similar to power-cycling the phone (powering it on and off). When a phone is restarted, it clears its current registration with Cisco CME and registers again. Restart is faster than reset. A restart is sufficient for configuration changes such as adding another line or speed dial to the phone. A reset is required for system-level changes such as upgrading a phone load or changing the URL for a directory search.

Cisco IP phones can be reset from a Cisco CME router console, from the Cisco CME GUI, or from the phone itself. A restart can be issued from the Cisco CME console or the Cisco CME GUI. You can reset or restart individual phones from the Cisco CME router console by issuing the reset or restart command, respectively, from the ephone configuration prompt.

You can reset from the phone itself by entering the key combination **#** from the phone dial pad. The phone should be in the on-hook state when you enter the key combination.

Erasing the Phone Configuration

In rare situations, you may have to erase a phone's current configuration. One such situation is when you physically move the phone from one Cisco CME system to another.

The running configuration of a Cisco 7960G, 7940G, or 7970G IP Phone can be erased from the phone. This provision is available on the Settings->Network Configuration menu. Scrolling through this menu using the rocker button or entering the number 36 takes you to the option erase configuration. If the phone is in an unlocked state, you should see a softkey labeled yes. Pressing the yes softkey gives you another softkey labeled save. If you select the save softkey, the phone's current configuration is erased, and the phone reboots.

Accessing Call Parameters for an Active Call

The Cisco 7940G, 7960G, and 7970G IP Phones have a provision to view call parameters such as the number of packets sent and received, the codec used for the current call, and delay and jitter readings for an active call. This is helpful when you're troubleshooting issues related to voice quality.

When a call is active, press the ? button twice to show the current parameters on the LCD screen. The parameters are updated dynamically. If there are two or more active calls on the phone, the display reflects the parameters for the call that is in the connected state.

Summary

In this chapter, you learned about the different phones that can be connected to a Cisco CME system and the options and features available for each device. It also included a discussion of the hardware capabilities, the key (button) operation, and the configuration of each phone.

The next chapter covers some of the important call processing features available on a Cisco CME system and how to configure them.

Chapter 5. Cisco CME Call Processing Features

This chapter covers the following topics:

- IP phones and lines
- Shared lines
- Hunt groups
- Intercoms
- Paging
- Line overlays
- Call pickup
- Softkey customization
- Call transfer and forward

This chapter describes the primary call processing features of Cisco CallManager Express (CME) and shows how you can combine them to produce an extensive set of call handling behaviors. It includes a basic discussion of the advantages of IP telephony for the small office and relates these to the more traditional time-division multiplexing (TDM) or analog-based telephone systems historically used in the small private branch exchange (PBX) and Key System marketplace.

This chapter explains the terminology and Cisco IOS commands (command-line interface [CLI]) used to configure IP phones, extension lines, shared lines, overlays, intercom, paging, call park and pickup, hunt groups, and other forms of call coverage. One of the key perspectives to understanding Cisco CME is that it is built on top of a Cisco IOS router. This means that the same modular feature approach that dominates the general Cisco IOS command-line organization is carried forward into the Cisco CME structure. The result is that individual component features are designed to be as modular and flexible as possible. It also means that it is often possible to combine features to produce some fairly complex operations. Some of these combinations are not obvious from a quick glance at the CLI. This chapter is intended to help you understand and use some of the available flexibility.

The sample configurations in this chapter are presented using the Cisco IOS CLI. Many of the configurations described can also be generated using the web browser graphical user interface (GUI). In both cases, the configurations generated are stored identically in the router's nonvolatile memory in CLI format. The CLI presentation is more compact and easier to grasp than an equivalent series of GUI screen shots. The CLI presentation also shows the integration of some Cisco CME-specific functions with the CLI commands for related but generic Cisco IOS router functions, because the generic Cisco IOS commands usually don't have a GUI equivalent. The CLI format is also convenient for many readers who may already be very familiar with the Cisco IOS CLI. The GUI is more extensively covered in Chapter 13, "Cisco IPC Express General Administration and Initial System Setup," and Chapter 14, "Configuring and Managing Cisco IPC Express Systems."

IP Phones and IP Phone Lines

IP phones may appear to be very similar in appearance to the digital phones used with a TDM-based PBX, at least on initial inspection. IP phones do behave in very similar ways for basic call operations. When you lift the handset, you hear dial tone. When an incoming call arrives, the phone rings. Phone users expect this behavior, which makes the introduction of IP phone technology as a replacement for traditional TDM-based telephony relatively painless for the vast majority of (nontechnical) phone users. In the case of traditional TDM-based telephony, the basis of the phone user interface is rooted in the physical structure of the typical digital TDM PBX. This in turn has its roots in the analog PBX systems that preceded it.

With IP telephony, some conscious and deliberate effort has gone into replicating the traditional phone user interface, because many of the historic engineering considerations that dictated design in the TDM PBX world are no longer applicable. This is well illustrated by considering the idea of "phone extensions" or "phone lines" for IP phones. In an analog PBX or Key System, the number of twisted-pair cables connected to the phone determines how many lines the phone has access to. If you want more phone lines, you have to add more wires. This is still mostly true for digital TDM phones. An example is a Basic Rate Inerface (BRI) phone with a twisted-pair cable carrying 2B + Dthat is, two bearer channels (audio) plus one data channel (signaling).

For an IP phone, there is no direct relationship between the physical wiring and the number of lines that an IP phone supports. IP phones based on 100-Mbps Ethernet connections can theoretically support hundreds of phone lines. How many lines an IP phone supports is instead determined solely by the design of the phone user interface, not the physical connectivity to the system equipment cabinet. The user interface might be a traditional looking one that has a dedicated physical button for each line the phone supports. Alternatively, the IP phone might simply have a touch screen. In this case, the number of square inches available on the display may determine the maximum number of lines accessible to the user. Other variations on user interface design might include the use of pull-down menus or scroll bars to select a phone line. The extreme example of this is the PC softphone. A softphone is simply an application program running on a PC where you select a phone line from the PC display with a mouse click.

The next section describes how Cisco CME deals with phones and phone lines.

Cisco CME Ephone and Ephone-dn

In the Cisco CME product, an IP phone device is called an ephone (short for Ethernet phone). The phone lines that are associated with the ephone are called ephone-dn (Ethernet phone directory number [DN]). An ephone-dn is made up of the following two subcomponents:

- Virtual voice port
- Dial peer

The virtual voice port is the nearest direct equivalent to a physical phone line in a Cisco CME system. The virtual voice port is the object that maintains the call state (on-hook or off-hook). The dial peer is the object that determines the phone number associated with the virtual voice port. A dial peer can do many additional things besides control the virtual voice port's phone number, such as apply translations to called and calling numbers. A virtual voice port can be associated with multiple dial peers and, therefore, can have multiple phone numbers associated with it.

Figure 5-1 shows that ephone-dn 7 creates, or is associated with, virtual voice port 50/0/7 and a plain old telephone service (POTS) dial peer that references virtual voice port 50/0/7. The dial peer contains the voice port's phone number. It is used for call-routing purposes for incoming calls. The virtual voice port contains the station ID that sets the caller ID properties (name and number) for the ephone-dn (used for outgoing calls).

Figure 5-1. Ephone-dn Components: Voice Port and Dial Peer

Implementing Shared Lines and Hunt Groups

Although it is widely used, the term hunt group seems to mean different things to different people and in different contexts. It's a general term that covers the distribution of calls among a group of phones. In this chapter, the term call coverage refers to the general concept of call distribution across phones. This helps avoid confusion with the Cisco CME CLI command ephone-hunt. The ephone-hunt command is used to configure a specific kind of call coverage that involves sequential selection among a group of selected phone lines.

Many other types of call coverage are addressed with other commands, combinations of commands, and configurations. Examples include the use of shared lines, overlay-dn, call forwarding, secondary ephone-dn numbers, and combinations of these.

This section describes a number of different types of call coverage:

- Simple sequential ringing of a selected set of phone lines
- Ringing of multiple phone lines at the same time, referred to here as parallel hunting
- Call forwarding from one line to another using call forward on busy or no-answer
- Combination sequences that involve sequential ringing of multiple phone lines, referred to here as sequential parallel hunting

Notice that all these refer to ringing phone lines, not phones. Calls are sequenced through an ordered set of phone lines. Which phones ring depends on the binding of the lines to the phones. Each line can be bound to multiple phones.

The following sections discuss several types of call coverage.

Cisco IOS Voice Dial Peer Hunting

A good place to start a discussion of call coverage for Cisco CME is to look at the voice dial peer hunting mechanism that is a standard component of the Cisco IOS voice infrastructure. Dial peers have a very large number of configuration options. The most basic options are related to matching a telephone number and directing the call to a specific voice port or VoIP destination. More advanced options relate to number manipulation (using translation rules or translation profiles), application selection (including custom TCL scripts), protocol options (such as fax relay and dual-tone multifrequency [DTMF] relay), voice codec selection, and caller ID manipulation and restriction.

Some dial peer operations relate to selecting a dial peer associated with the call's originator (incoming dial peer matching) and selecting a dial peer associated with the call destination (outgoing dial peer matching). Multiple parameters can affect the selection of the incoming and outgoing dial peers, but the primary matching criteria are the called and calling party numbers.

Describing all the dial peer options in detail is outside the scope of this book; however, numerous books deal with this subject in more depth, including the following:

- Deploying Cisco Voice over IP Solutions by Jonathan Davidson (Cisco Press, 2001).
- Cisco Voice over Frame Relay, ATM, and IP, edited by Steve McQuerry, Kelly McGrew, and Stephen Foy (Cisco Press, 2001).

Creating an Intercom

Cisco CME supports single-button push-to-talk and push-to-respond intercom lines. You can create an intercom arrangement between any two (multiline) IP phones that support speakerphone operation. You can even operate an intercom across a VoIP connection using either SIP or H.323. Cisco CME's intercom function is built using two functions:

- Autodial at the initiating end of the intercom
- Autoanswer-with-mute at the receiving end

To create an intercom you assign a line button on each of the two phones to operate as an intercom line. Pressing the intercom line button selects the line and triggers the autodial function toward the second phone. The receiving phone receives the incoming intercom call on its intercom line. This line autoanswers the call and activates the phone in speakerphone mode and sounds a beep. It also forces the speakerphone to mute to protect the privacy of the intercom recipient. The audio path is open from the initiator to the receiver. To respond to the intercom, the recipient simply presses the mute button to unmute the audio path back to the originator.

The intercom Command

Example 5-21 shows a configuration of an intercom between two IP phones.

Example 5-21. Intercom Lines

```
router#show running-config
ephone-dn 1 dual-line
   number 1001
   name John Smith
ephone-dn 2 dual-line
   number 1002
   name Jane Smith
ephone-dn 3
   number 1111
  intercom 1112 label Jane
ephone-dn 4
   number 1112
  intercom 1111 label John
ephone 12
   mac-address 000d.1234.0efc
   button 1:1 2:3
ephone 15
   mac-address 000d.5678.0dcf
   button 1:2 2:4
```

Example 5-21 shows two phones. John's phone has button 1 as his primary extension line. Button 2 on John's phone is an intercom line. This line is set to autodial Jane's phone using the number 1111. The button is labeled "Jane" to show that pressing the button intercoms to Jane.

Note

The intercom lines are configured in the default single-line mode.

Jane's phone is configured to match to John's phone. Button 1 on Jane's phone is her primary line. Button 2 is set to autodial John's phone and shows the label "John" next to the button. The default configuration for an intercom line is

Using Private Lines

Similar in some ways to the courtesy phone application for the intercom command, Cisco CME 3.2 introduces the FXO trunk command. This can be used to provide an emulation of a dedicated private PSTN line for a specific phone user. This allows Cisco CME to create the user appearance that one of the buttons on a Cisco IP phone is directly connected to a specific PSTN subscriber line (usually a dedicated FXO port connected to a specific PSTN phone number). One potential application for this is in a bank branch office where the bank manager has an internal extension number for regular calls (perhaps forwarded to the manager by a receptionist) plus a direct private line used for important clients. The private line also has voice mail service provided by the PSTN. This provides message waiting indication (MWI) by way of a stutter dial tone. To hear the MWI indication, the manager selects the private line and hears dial tone provided by the PSTN.

Example 5-25 shows how the trunk command is used to create a private line on an IP phone.

Example 5-25. FXO **trunk** Command

```
router#show running-config
voice-port 1/0/0
  connection plar-opx 1082
dial-peer voice 82 pots
  destination-pattern 82
    port 1/0/0
ephone-dn 10
    number 1010
    name manager
ephone-dn 11
  number 1082
    name private-line
  trunk 82
ephone 1
    button 1:10 2:11
```

Example 5-25 shows ephone 1 configured with two lines. Button 1 is a normal extension (number 1010) using ephone-dn 10. Button 2 is the private line using ephone-dn 11. Incoming PSTN calls on FXO port 1/0/0 are routed directly to ephone 11 (extension number 1082) by the connection plar-opx 1082 command that is shown under voice-port 1/0/0.

Outgoing calls on the private line are routed to voice port 1/0/0 by the trunk 82 command within ephone-dn 11. This causes all calls dialed on ephone-dn 11 to have the digits 82 prefixed to the dialed number. The 82 prefix causes the calls to match to the destination-pattern 82 in dial peer 82 and, thus selects voice port 1/0/0 for the outbound call.

When the phone user dials the number, such as 555-0510, the trunk command prefixes the digits 82 to create the number 825550510. The leading digits 82 match the destination pattern for the voice port dial peer. The leading 82 digits are stripped off, and the remaining digits, 5550510, are forwarded to the PSTN line. Note that in some cases, you may need to adjust the time delay before the digits are passed to the PSTN line to avoid their being sent before the PSTN line is ready to accept them. You can do this using the prefix command (under the POTS dial peer) and using commas to insert 1-second units of delay. For example, the command prefix,, inserts 2 seconds of delay.

The appearance that the phone button is directly connected to the FXO port's PSTN line is obtained by means of the one-to-one association that's created by the combination of the connection plar-opx binding of the FXO port to the ephone-dn, plus the trunk binding of the ephone-dn back to the FXO port. Because of this arrangement, whenever the FXO port is in use, it follows that the ephone-dn is also in use. Therefore, the ephone-dn's status reflects the FXO port's status.

To maintain this direct one-to-one binding, call operations that could break the one-to-one binding are disabled by the trunk command. This means that functions such as call transfer and call forwarding are not supported when the trunk configuration is used. However, functions such as call hold and conference are supported, including the ability to

Paging

You can set up a Cisco CME system to provide audio paging using the speakers of your IP phones to broadcast the paging audio output. This feature works in conjunction with IP phones that have a speakerphone mode. Only IP phones that are idle are used to output paging audio. IP phones can still be used to make or receive calls during paging. When the phone is used, it simply drops out of the page.

You can create paging groups or zones that output paging audio only on specific groups of IP phones. You can also combine multiple paging groups to output audio paging to multiple paging groups at the same time.

Using your IP phones to provide an audio paging system can save the additional cost of installing a separate overhead audio paging system. If you already have a conventional overhead audio paging system, you can also use this with Cisco CME. You simply need an available physical voice port on your router that can connect to the paging system. Ear and mouth (E&M) voice ports are the easiest to use because they do not usually require external adapter hardware. Because E&M ports usually come in pairs, you can also use the second E&M port as an input to connect to an external music on hold audio source.

The following sections describe how you can use the Cisco CME paging features.

Paging Groups

To configure an IP phone-based paging group, you first set up an ephone-dn entry in your system to act as the pilot number for the paging group. Like most of the special-purpose ephone-dns you create for your Cisco CME system, the paging ephone-dn is not directly bound to any of your IP phones. You don't use this ephone-dn in a button command.

The paging group pilot number is the number you dial from a phone to output audio to the paging group. A sample configuration is shown in Example 5-26. Note that you can dial into a paging group from any phone, including via a VoIP connection. You can also set up speed-dial buttons on your IP phones to dial into the paging number for one-button push-to-page operation.

Example 5-26. Paging Pilot Number Configuration

```
router#show running-config
ephone-dn 14
   number 6112
   name Sales
  paging ip 239.1.1.1 port 2000
```

The paging-dn is set up with a pilot number of 6112. When you make a call to 6112, it is answered by the paging-dn. The audio stream from your phone to the paging-dn is broadcast to all the phones in the paging group for ephone-dn 14 using IP Multicast on address 239.1.1.1 and port 2000.

As soon as you have created your paging-dn, you then create a group of IP phones to include in your paging group, as shown in Example 5-27. You do this using the paging-dn command within the ephone command mode. Set the paging-dn command in the ephone command mode to select the tag number of the ephone-dn that you configured as a paging-dn (14 in this example).

Example 5-27. Paging on IP Phones

```
router#show running-config
ephone 12
   mac-address 000d.1234.0efc
   button 1:1 2:6
  paging-dn 14
ephone 15
```

Implementing Overlays

The Cisco CME overlay feature provides a way to work around the physical button limits on your IP phones. Instead of a normal one-line-to-one-button mapping arrangement, you can map up to ten lines or ephone-dns to the same physical phone button. This allows you to use the same phone button to answer incoming calls on any of the up to ten ephone-dns associated with the button.

Note

Although you can map ten ephone-dns to the same button, you can work with and see only one ephone-dn at a time.

Earlier, you saw one usage (in Example 5-17) for overlay-dns, where it allowed access to multiple instances of ephone-dns that have the same telephone number. See Figure 5-2 earlier in this chapter.

The Purpose of an Overlay-dn

An overlay-dn associates or binds from two to ten ephone-dn IP phone lines onto a single IP phone button (even on single-line IP phones). You can use separate overlay-dn arrangements on each separate IP phone button. Each IP phone can use an independent set of ephone-dns for overlay for each of the phone's buttons.

An overlay-dn acts as a multiplexor. It dynamically selects the most appropriate ephone-dn to present on an IP phone button from within the configured overlay-dn set. When you receive incoming calls, the first ringing ephone-dn in the overlay set is presented. When you make an outgoing call, the first idle ephone-dn in the overlay set is selected.

You can configure the ephone-dns used in an overlay set as either single line or dual line. However, all the ephone-dns in the same overlay set must be of the same type (single or dual line).

Using Overlay to Overcome Phone Button Count Limits

The simplest use of overlay-dn is to overcome the limited number of physical buttons available on an IP phone. In a simple Key System case where you have four incoming PSTN trunk lines and four IP phones, you can make each line available on all phones simply by using one button per line. You can do this using a Cisco 7960 IP Phone, which has six buttons. With this arrangement, you can answer any of the four incoming lines on any of the four IP phones.

However, you cannot use this simple one-button-to-one-line mapping if you want to have ten incoming PSTN lines and ten IP phones (assuming a six-line Cisco 7960 IP Phone). There simply aren't enough line buttons to do this (unless you add a Cisco 7914 IP Phone Expansion Module to your Cisco 7960 IP Phones).

Example 5-30 shows how you can map ten incoming PSTN lines to a single button on an IP phone.

Example 5-30. Overlay-dn Configuration

```
router#show running-config
ephone-dn 101
   number 4085550101
   no huntstop
ephone-dn 102
   number 4085550101
   preference 1
   no huntstop
ephone-dn 103
   number 4085550101
   preference 2
   no huntstop
ephone-dn 104
   number 4085550101
   preference 3
```

Invoking Call Pickup

The call pickup feature allows you to retrieve calls on hold in a park slot and to move calls from one phone to another. You invoke the call pickup feature from the phone you want to move the call to. You can move calls that are in either the (incoming) ringing state or the call hold state. To invoke call pickup, simply press the pickup softkey on the IP phone, and enter the extension number of the ephone-dn that has the call you want to move.

Call pickup is also often used in conjunction with paging. In this case, you place a call on hold, either at an extension or in a park slot, and then use the paging system to request that a coworker pick up the call you just parked. The park-and-pickup combination is a popular feature used in the retail store environment.

Pickup of a Ringing Extension

You can use the pickup softkey to move a call on a ringing extension on another phone to your phone. When you invoke call pickup, the Cisco CME system triggers a call forward from the phone with the ringing extension to your phone. Because the call pickup for a ringing extension uses the call forwarding mechanism, the general restrictions and considerations that apply to forwarded calls (forward on no-answer) also apply to calls for pickup on ringing. This means that the max-redirects limit set under telephony-service may affect the call pickup feature.

Pickup of a Call on Hold

You can use the pickup softkey to move a call that is on hold at an extension on another phone and move it to your phone. When you invoke call pickup, the Cisco CME system triggers a blind call transfer from the phone with the on-hold call to your phone. Because the call pickup for an on-hold call uses the call transfer mechanism, the general restrictions and considerations that apply to transferring calls also apply to calls for pickup on hold.

Pickup Groups

The group pickup feature operates in a manner similar to simple extension pickup. To use the group pickup feature, press the gpickup softkey and enter a pickup group number. You can use the group pickup feature to pick up any ringing call in a designated group of phones. You cannot use the group pickup feature to pick up calls that are on hold.

You can assign each phone in your Cisco CME system to a pickup group. You can have any number of pickup groups in your Cisco CME system.

If you create only a single pickup group in your Cisco CME system, you can pick up calls from within the single pickup group by simply pressing the gpickup softkey. If your Cisco CME system has only a single pickup group, there is no need to enter a pickup group number.

If you have more than one pickup group, you can pick up calls from within a phone's local pickup group by pressing the gpickup softkey followed by the star (*) key. This provides a shortcut instead of requiring you to enter the phone's own pickup group number.

To assign a phone to a specific pickup group, use the pickup-group command within the ephone configuration command mode, as shown in Example 5-32.

Example 5-32. Pickup Group Configuration

```
router#show running-config
ephone 1
   mac-address 000d.1234.ecfd
   button 1:10
   pickup-group 201
```

Customizing Softkeys

Cisco CME 3.2 and later allow you to customize the set of softkeys displayed to the phone user in each stage of a phone call. For example, by default in the connected call state, a Cisco 7960 IP Phone shows the following six softkeys:

- Acct (account code entry)
- Confrn (three-party conference)
- Endcall
- Flash (sends a hookflash signal to the PSTN line)
- Hold
- Trnsfer (call transfer)

Because a Cisco 7960 IP Phone has only four physical softkey buttons, the phone displays a more softkey as the rightmost button to allow the phone user to scroll to access all the keys. Because the more softkey itself uses one of the physical buttons, the user sees the feature softkeys in two sets:

- Hold
- Trnsfer
- EndCall
- More
- Confrn
- Acct
- Flash
- More

The softkey customization feature allows you to change this default behavior, and choose the set of keys you want to display. It also allows you to control the order of the keys so that you can move the more frequently used keys to the first page. For example, you may choose to remove the Acct and Flash keys from the connected state key set, leaving just the following four keys:

Configuring Call Transfer and Forward

Configuring call transfer and forwarding for H.323 VoIP calls is a fairly complex task in most real-world H.323 VoIP networks. This is especially true if you have a mixture of H.323 VoIP systems from different vendors. Even if you have an all-Cisco H.323 VoIP network, there are still interactions to consider, unless all your VoIP systems are running relatively up-to-date Cisco IOS software. This means having at least Cisco IOS 12.3 software in all your voice-enabled routers. Ideally, you should have Cisco IOS 12.3(4)T or later software (this is the Cisco CME 3.0 base code version). There are also some special considerations if you are using a Cisco CallManager in addition to your Cisco IOS-based Cisco CME systems, which you'll learn more about in Chapter 7, "Connecting Multiple Cisco CMEs with VoIP." The good news is that with the right software and configuration, there are workable solutions for most of your VoIP call transfer and forwarding needs.

In a VoIP network, getting an optimized system working for call transfer and forwarding requires the active cooperation of all endpoints involved in a call transfer or call forward. This means your ability to perform a call transfer or forward depends on the capabilities of the calling party's VoIP system as well as your Cisco CME system configuration. A call transfer also depends on the capabilities of the final VoIP system that you are transferring the call to.

In traditional TDM-based PBX telephony, call transfer and forwarding usually operate within the limited scope of a single PBX system and, therefore, are simpler operations. For example, you are often limited to call transfers between extensions on the same PBX only.

In a VoIP-based system, you can potentially transfer or forward calls between any VoIP endpoints, regardless of their physical location. Of course, being able to do this in practice requires making sure that you have support for transfer and forwarding built into all your VoIP endpoints.

With Cisco CME, you have three basic choices for the protocol used to support call transfer and forwarding for H.323 VoIP calls:

- Standards-based H.450 Strongly recommended because it provides for optimal call paths and unlimited sequential transfers and forwards
- Cisco-proprietary H.323 extension Mostly obsolete, but useful if you are using software older than Cisco IOS 12.2(15)T
- Hairpin call routing Maximum compatibility but uses more WAN bandwidth and results in higher delay and jitter

The default H.323 call transfer protocol used by Cisco CME is the Cisco-proprietary mechanism. This mechanism supports only blind call transfer (that is, no transfer consultation). It is selected as the default simply for purposes of backward compatibility with earlier Cisco IOS versions.

The default call forwarding mechanism provides for automatic local forwarding only (that is, within the same Cisco CME system). It does not provide forwarding display update notification of the call forwarding to the calling party's IP phone. For incoming VoIP calls from another Cisco CME system that are nonlocally forwarded to a third Cisco CME system, the Cisco-proprietary H.323 protocol extensions are used.

Even if you do not require H.323 VoIP call transfers (because you do not need to make calls across an IP connection to another site), you should still select the H.450 configuration method for call transfers. This enables call transfer with consultation for local calls within your system and for PSTN calls that use PSTN voice ports that are physically on your Cisco CME router. (PSTN voice ports on a router other than the Cisco CME system appear as H.323 VoIP calls to the Cisco CME system.) It also prepares your system to use the standards-based H.450 protocol in case you want to add support for H.323 or SIP VoIP transfer and forwarding to another site at some

Summary

In this chapter, you read about some of the more popular Cisco CME phone and call processing features. You saw examples of how these features can be configured and combined to provide a rich and flexible set of functions. You also saw how to configure call transfer and forwarding functions in a variety of network scenarios.

Chapter 6. Cisco CME PSTN Connectivity Options

This chapter covers the following topics:

- Cisco CallManager Express (CME) analog and digital Public Switched Telephone Network (PSTN) interface hardware, configuration, and features
- PSTN call switching and routing options
- Digit manipulation tools to translate phone numbers
- PSTN trunk failover options

In previous chapters, you learned about the Cisco IP Communications (IPC) Express platforms and architecture, Cisco IOS voice infrastructure software features, and configuring call routing features between endpoints such as IP phones and Public Switched Telephone Network (PSTN) interfaces.

This chapter focuses on the hardware options, configurations, and detailed features that Cisco IPC Express offers for connecting your business to the PSTN. This PSTN connectivity is not particular to Cisco IPC Express but is part of the generic voice-enabled feature set of the Cisco router platforms.

Router PSTN connectivity is generically referred to as voice gateway functionality, offering a gateway for voice over IP (VoIP) calls to, and from, traditional analog or digital PSTN or private branch exchange (PBX) calls. You can use a router voice gateway to connect to PSTN central office (CO) switches, private branch exchanges (PBXs), Key Systems, time-division multiplexing (TDM)-based interactive voice response (IVR) systems, traditional TDM-based voice mail systems, and any other legacy (non-IP) voice processing or telephone equipment.

This chapter explores several aspects of Cisco CME connectivity to the PSTN, including the following:

- Standards-based telephony signaling systems and protocols supported by Cisco IOS, which, in turn, determine what traditional TDM or analog systems you can connect to and what features you get when using this type of connection
- A brief overview of the Cisco voice gateway hardware choices and the voice port densities and features they provide
- Sample Cisco IOS configurations for different types of PSTN connections
- Network design and call switching considerations for connecting to the PSTN from your IP network

Traditional telephony terminology is used throughout this chapter. It has a more precise meaning here than in other chapters, because the topic of discussion is connecting a traditional telephony system, the PSTN. If you are unfamiliar with the telephony meanings of terms such as trunks, lines, caller ID, direct inward dial (DID), and dialed number information service (DNIS), refer to the glossary to review these terms.

Trunk Signaling Systems

Cisco IOS PSTN connectivity complies with the relevant standard signaling systems used by the PSTN and other telephony-switching systems. Cisco IOS routers support all the signaling variations in general use in the world today. No matter where your business is located, you should be able to connect easily to the PSTN with the analog or digital signaling options described in this section.

Analog Signaling

Low-density PSTN connectivity typically implies an analog connection. In some geographies Basic Rate Interface (BRI) is used instead, as discussed in the "Digital Signaling" section. Analog signaling is also used for connections to analog stations (such as fax machines and traditional analog phones). Table 6-1 summarizes the analog signaling variations supported by Cisco IOS voice gateways.

Table 6-1. Analog Signaling Support by Cisco IOS

Signaling	Description	Typical Use
FXS	Foreign Exchange Station	Used to connect to analog phone sets or fax machines. Occasionally also used to connect to a PBX or Key System if it offers only FXO interfaces.
FXO	Foreign Exchange Office	Generally used to connect to an analog PSTN line. Also used to connect to a PBX or Key System FXS interface. Can be connected to any interface where a standard analog phone is currently connected.
E&M	Ear and Mouth	Used to connect to an analog PBX.
Analog DID	Analog Direct Inward Dial	Used to connect to an analog PSTN line that has DID service on it.
CAMA	Centralized Automatic Message Accounting	Used to connect to the PSTN for emergency services (911 calls) in North America.

To connect your Cisco CME system to the PSTN for normal analog business line service, you use FXO interfaces. FXO ports, like all the other analog interfaces, carry one call per port, so each RJ-11 port on your Cisco CME router connects to one line from the PSTN and carries a single call at a time. (A second call gets busy tone if it tries to use the same port or line.)

Note

Note that on voice interface cards such as the NM-HDA and EVM-HD-8FXS/DID, which contain a single RJ-21 50-pin connector, the individual analog ports carried in the single cable are broken into separate RJ-11 ports by a break-out box.

Cisco IOS PSTN Telephony Interfaces

You can add numerous modular cards to your Cisco CME router to support PSTN connections of the types discussed in the preceding section. These technologies and hardware cards are not particular to Cisco CME. They can be used on any Cisco router platform that supports the card in question, independent of whether Cisco CME is enabled on the router. For example, you may choose to have two separate routers in your officeone configured for Cisco CME and the other as the PSTN voice gateway. Or you can combine both functions in the same router.

The following sections cover these hardware choices in greater detail, first looking at analog and then at digital PSTN trunks.

Analog Trunks

Voice interfaces range from two- and four-port FXO/FXS/E&M/DID cards up to 96/120-channel quad T1/E1 interfaces. The physical telephony interface for analog and BRI ports is provided by a plug-in voice interface card (VIC) and for a T1/E1 port by a voice or WAN interface card (VWIC).

Using various combinations of VICs and VWICs on a Cisco IOS router, you can build a Cisco CME system that includes a range of physical telephone interfaces. You can assemble a small analog telephony system with a few FXO ports used to connect to PSTN subscriber lines, or you can use digital telephony interfaces such as T1/E1 and ISDN BRI/PRI, or any combination of these. The specific hardware cards offering analog trunk and station (analog phone or fax machine) interfaces are discussed next.

Analog Trunk and Station Hardware

The analog interface cards listed in Table 6-3 are used to provide low-density analog PSTN interfaces. VICs are placed in a WAN interface card (WIC) slot (supported on the Cisco 1751 and 1760), in a high-speed WIC (HWIC) slot on the router (supported on the Cisco 2800 and 3800 series), or inside a network module (Cisco 2600, 2800, 3700, and 3800 series) such as the NM-HD-1V, NM-HD-2V, NM-HD-2VE, or NM-HDV2. For high-density analog PSTN interfaces, the NM-HDA (supported on the Cisco 2600, 2811, 2821, 2851, 3700, and 3800 series) or the EVM-HD-8FXS/DID card (supported on the Cisco 2821, 2851, and 3800 series) can be used.

Table 6-3. Analog Interfaces, Signaling, and Density

Interface Card	Signaling	Density
VIC-4FXS/DID	FXS and analog DID	Four ports
VIC2-2FXO	FXO and CAMA	Two ports
VIC-2DID	Analog DID	Two ports
VIC2-4FXO	FXO and CAMA	Four ports
VIC2-2FXS	FXS	Two ports
VIC2-2E/M	E&M	Two ports
NM-HDA-4FXS, EM-HDA-8FXS, and EM-HDA-4FXO	FXS and FXO	Four ports on the baseboard, but can be expanded up to 12 FXS ports by adding an EM-HDA-8FXS card to the network module (NM), or up to

PSTN Call Switching

The preceding sections explored the various PSTN trunk types, signaling methods, and router hardware you can use to connect to the PSTN. But there are more considerations than just physical connectivity. One thing to keep in mind is that the PSTN numbers and your internal extension numbers are almost certainly not the same, at least not the same length. Digit translation must occur to map one set of numbers to another. You'll learn more about this in the later section "Digit Manipulation."

Another consideration is what PSTN numbers (and how many) your business has or needs. Do you have just one main office number, and the receptionist directs all calls to the correct employee? Do you need an AA menu to have callers switch to the person or service they want to speak to? Should you have DID numbers for all or some of your employees? Do you prefer a Key System type of operation, where a series of PSTN numbers appear as distinct line appearances on a number of phones, and any employee can pick up any call? There is interaction between your business needs, the PSTN service you get from your provider, the capabilities of the physical connection to the PSTN, and the Cisco CME configuration (IP phone button appearances) to use.

The physical connection is likely dictated in large part by cost, your office's geographic location, and the number of voice channels your office needs. You may desire DID service for your business. However, if it's not offered in your area at a cost-effective level, you have little choice but to settle for non-DID service.

The following sections explore considerations about how calls may be routed depending on whether you have DID service and, even if you do, how you can handle calls to non-DID destinations within your business.

PSTN Call Switching with DID Enabled

Many offices deploying Cisco CME have DID capability from the PSTN provider for some subset of employees. PSTN calls to DID destinations can be switched automatically to the employee's phone without any manual intervention.

There are two situations to consider, depending on how DID numbers are allocated to destinations within your business:

- All employees, the AA, and voice mail pilot numbers have DID numbers assigned. In this configuration, PSTN calls can be switched as follows:

  The main office number (non-DID calls) terminates on the AA pilot (for AA assistance) or on an IP phone extension (for receptionist assistance).

  Employee DID numbers terminate on the extension for that person's IP phone.

  The number employees call from PSTN locations to check their voice mail terminates on the voice mail pilot number.

- Some employees have DID numbers assigned, and others do not. In this configuration, PSTN calls are handled as follows:

  Calls to the main office number and for PSTN voice mail checking are handled as per the preceding scenario.

  Calls to employees with DID numbers terminate on the extension for that person's phone.

Digit Manipulation

There are various reasons to manipulate the digits dialed by the caller on a voice system. The most common reason is to allow both internal calls (from other extensions) and external calls from the PSTN (where a full E.164 phone number is delivered) to terminate directly on the user's phone without needing a receptionist to intercept and redirect the call.

Note

E.164 is an International Telecommunication Union (ITU) specification that describes international telephone dial plans. It specifies phone number attributes such as international dialing codes, regional (area) codes, and the minimum and maximum length of each field in the phone number. Voice systems use the E.164 specification to parse and interpret phone numbers. PSTN numbers are always fully qualified E.164 numbers, whereas extensions within your business typically aren't, because they are private numbers of local significance only.

Here are some other reasons to translate (or manipulate) digits:

- To allow IP phone users to call each other directly by extension, and also to access the PSTN
- To allow for site access codes in a multisite on-net dial plan and to strip these digits to extract the extension as soon as the destination site is reached
- To allow for variable-length external (off-net) dialing while maintaining fixed-length internal dialing
- To block calls to certain numbers
- To redirect calls to certain numbers

For example, suppose your employee, Grace, is at extension 3001, and her PSTN DID number is 444-555-3001. Without some form of digit manipulation or live intercept, a call incoming from the PSTN that dialed 444-555-3001 will not match the ephone-dn definition for Grace's phone, which contains only her extension, 3001. Therefore, a method is needed to translate the string 4445553001 to 3001.

Several Cisco IOS digit manipulation tools can translate phone numbers. The following are the most common:

- Dial peer commands
- Cisco CME dialplan-pattern command
- IOS translation rules

Dial Peer Commands

You can include several commands on a POTS dial peer to add, suppress, or substitute the digits forwarded to the PSTN trunk interface:

-

PSTN Trunk Failover

Larger offices that use a digital trunk, such as a PRI, often need a backup method to connect to the PSTN. This requirement results in the PRI being the main PSTN connection point in addition to a handful of FXO trunks (typically to back up a T1) or BRI interfaces (typically to back up an E1) used if the main interface is down.

In this configuration, the dial peers directing calls to the main interface must be duplicated to also point to the backup interface. You can force calls to use the main interface when it is available by using the preference command on the dial peers pointing to these trunks. The use of the preference command is covered in Chapter 5, "Cisco CME Call Processing Features."

Another need is to have a backup mechanism for a small office with FXO trunks if a power failure should occur. FXO hardware supports a feature called FXO Power Failover that allows a hardware (relay) connection between a red phone (a specially dedicated analog telephone in your office that normally is not in use) and the PSTN line, in case the router is not powered.

On Cisco voice hardware, the NM-HDA-4FXS FXO expansion card (the EM-HDA-4FXO) and the EVM-HD-8FXS/DID FXO expansion card (the EM-HDA-6FXO) each have one port per card that has this power failover capability. Other Cisco FXO hardware cards do not support this feature.

Summary

In this chapter, you learned how to connect your Cisco CME system to the PSTN. You reviewed the different signaling mechanisms and the features each provides to your business. You also learned about the Cisco IOS voice-enabled router hardware you can choose from to connect to the PSTN.

The latter part of this chapter focused on PSTN call switching and routing considerations to help you set up and optimize your office to handle PSTN calls in the manner that best fits your business model.

Chapter 7. Connecting Multiple Cisco CMEs with VoIP

This chapter covers the following topics:

- General considerations when integrating Cisco CallManager Express (CME) in H.323 and Session Initiation Protocol (SIP) voice over IP (VoIP) networks
- Integrating Cisco CME into an H.323 network with or without a gatekeeper
- dual-tone multifrequency (DTMF) relay for H.323
- Call transfer and call forwarding within an H.323 network using H.450
- Integrating Cisco CME into a SIP network

This chapter describes the ways in which you can use Cisco CME as a component of a larger network using the two major voice over IP (VoIP) protocolsH.323 and SIPto link multiple Cisco CME systems. It examines some of the considerations that apply within a networked environment that do not arise in simpler standalone configurations. This chapter focuses on the call handling implications of using Cisco CME in a network. The administration of networked systems is covered in Chapter 14, "Configuring and Managing Cisco IPC Express Systems."

Considerations When Integrating Cisco CME in H.323 and SIP VoIP Networks

H.323 is the dominant protocol deployed for VoIP networks from an installed-base perspective. Because H.323 is more mature than SIP, you can expect to see increased real-world interoperability between different vendors' H.323 products, particularly with basic call handling. However, many of the high-level VoIP networking considerations that apply to H.323 apply equally in the SIP context. Some technical and protocol-specific differences exist between H.323 and SIP VoIP networking, but for the most part, you'll find more commonality than difference, at least at the level of technical detail that this chapter addresses.

The shared aspects of the two protocols means that the overall high-level architecture and distribution of hardware and primary component roles within your VoIP network don't significantly depend on which protocol you choose to use for intersite VoIP. For networks built on either H.323 or SIP, you are dealing with peer-to-peer communication between sites. Therefore, you also need some kind of telephone number directory system to be able to resolve the IP address of the appropriate destination VoIP peer device for intersite calls.

In contrast, this similarity between H.323 and SIP does not extend to Media Gateway Control Protocol (MGCP) (and also Skinny Client Control Protocol [SCCP]), which takes a significantly different approach to telephony. Of course, it is still possible to connect Cisco CME to MGCP networks, primarily using either H.323 or SIP. Many MGCP Call Agent implementations (using MGCP internally for phone control) use H.323 or SIP to connect separate Call Agents (as intersystem peer-to-peer). Cisco CME itself does not support control of MGCP endpoints. Cisco CME uses SCCP for phone control, and SCCP shares many common traits with MGCP.

The term VoIP here specifically describes "long-distance" VoIP telephone calls that traverse a WAN. This interpretation excludes SCCP used to control local IP phones. Although SCCP technically does use VoIP technology, it is primarily used in the context of operating voice calls within the confines of a LAN with more or less unlimited bandwidth and many fewer concerns about security.

You can view the H.323/SIP versus SCCP contrast as the difference between interbranch office voice traffic and intrabranch office voice traffic, or alternatively as long distance (WAN) versus local VoIP (LAN). This division is useful in many ways, because it inherently supports the often-necessary difference in treatment of calls between internal and external phone users.

In some cases, you will want to treat H.323 calls as internal calls and won't want a high degree of differentiation in the treatment of LAN versus WAN calls, such as calls between separate systems on two floors of the same building. Cisco CME has features that address this, although currently you cannot treat a network of many Cisco CME systems as if they are a single logical entity with full intersite feature transparency. Both H.323 and SIP still have obstacles to overcome before this is really possible. Not least of these are issues surrounding meaningful interoperability with non-Cisco devices for services beyond basic calls.

When you extend VoIP calling into the WAN space, you might also have to consider the difference between VoIP calls that come from other Cisco CME nodes within your WAN network versus VoIP calls that are from VoIP Public Switched Telephone Network (PSTN) gateways or even from other independent external/wholesale VoIP carrier networks. You can link independent VoIP networks together and into your corporate VoIP network using IP-to-IP gateways. This arrangement may be desirable if you want to obtain international and long-distance phone service directly from a carrier-class VoIP service provider and have this linked at the VoIP level to your private enterprise VoIP network.

SIP potentially has some advantages over H.323 in terms of separating intersite VoIP calls from true external VoIP calls, because SIP uses the Internet concept of domains. It's a fair assumption that all of the intersite calls will use the same root domain name and that this fact can be used to make the required distinction. However, from a purely practical security point of view, most likely you will want any truly external VoIP traffic entering your corporate VoIP network to pass through an IP-to-IP gateway and also a firewall, regardless of whether you choose to use SIP or H.323. This means that you should have the opportunity to appropriately classify and mark the external calls at the point of entry in either type of network.

Integrating Cisco CME in an H.323 Network

There are two basic approaches to connecting a Cisco CME system to an H.323 network: the first uses no gatekeeper (GK), and the second does. A direct interconnection of sites with H.323 implies that each site must be knowledgeable about how to reach every other site. This works well in small networks of only a handful of nodes, but as the network grows larger, the configuration becomes increasingly cumbersome to maintain. In its simplest form, a gatekeeper is a device that provides a directory service that translates a telephone number into an IP address. Using a gatekeeper provides significant scalability by centralizing the interconnection of the individual sites so that each site needs to be aware of only the gatekeeper and not every other site in the network.

The following sections discuss different approaches to building Cisco CME networks. Rather than being alternative approaches, they represent a simpler approach for smaller networks with only a few nodes and a more scalable approach for larger multinode networks.

A Simple Two-Node Topology with H.323

In the simplest case, you can just connect two Cisco CME systems via an IP-enabled serial data link (or Ethernet), and configure VoIP dial peers on each system to symmetrically direct calls that are destined for nonlocal extension numbers to the other Cisco CME system. In other words, if the Cisco CME recognizes that the extension number being dialed isn't present in its internal list of phone numbers, it can assume that it should send the call to the other Cisco CME, as shown in Figure 7-1.

Figure 7-1. Simple Two-Node Cisco CME H.323 Network

Example 7-1 shows the relevant configuration extracts of the two systems. It shows a pair of Cisco CME systems that have extensions 1000 to 1099 on CME 1 (IP address 10.1.1.1) and 2000 to 2099 on CME 2 (IP address 10.1.2.1).

Example 7-1. Dial Peers for Figure 7-1

```
CME 1:
Router#show running-config
dial-peer voice 2000 voip
     destination-pattern 20..
     session target 10.1.2.1
     dtmf-relay h245-alphanumeric
     codec g729r8


CME 2:
Router#show running-config
dial-peer voice 1000 voip
     destination-pattern 10..
     session target 10.1.1.1
```

DTMF Relay for H.323

Dual-tone multifrequency (DTMF) relay is a mechanism for reliably carrying DTMF digits across VoIP connections. If you need to signal the 0 to 9, *, and # keypad digits (DTMF digits) from your IP phone across your VoIP network, you must configure DTMF relay. DTMF digits are also sometimes called TouchTone digits.

You should configure DTMF relay if you want to operate a remotely connected voice mail system, use calling card access for PSTN calls placed through a remote VoIP PSTN gateway, or access any kind of DTMF-driven interactive voice response (IVR) system (for example, telephone banking or airline flight information services).

DTMF Digits

DTMF describes a method of encoding telephone digits using two audio tones. For a conventional telephone keypad in which the keys are arranged in three columns by four rows, the first audio tone selects the row of the key, and the second audio tone selects the column. Each row-and-column tone uses a different audio frequency (pitch). This method of telephone digit signaling replaced the old-fashioned loop disconnect (dial pulse) digit dialing used by old rotary-style analog phones.

There are 16 DTMF digits (arranged as four columns by four rows). In addition to the standard 12 keypad digits0 to 9, *, and #an additional four digits form an extra fourth column of digits called simply A, B, C, and D. Because the ABCD digits are unavailable on a normal phone keypad, you are unlikely to ever come across these for normal phone calls. They are used occasionally by voice mail systems to operate an intersystem exchange of voice messages between separate voice mail systems using a standard called Analog Message Interchange Standard (AMIS).

Some security-type phones also use the ABCD digits for initial negotiation. You may also see these used in some Cisco CME configuration examples where there is a need to create telephone numbers that can't be directly dialed from a phone keypad. One example of this is if you want to create nondialable phone numbers for intercoms. You normally place an intercom call by pressing a button specifically configured for intercom (this works somewhat like a speed-dial button), so you don't need to be able to enter individual dialed digits.

Transporting DTMF Digits Reliably Using DTMF Relay

In the simple case of analog phones, the phone keypad digits result in the generation of DTMF audio tones. DTMF signaling works fine for analog phones connected directly to PSTN analog subscriber lines that travel only a relatively short distance to reach a central office (CO) telephone exchange. However, when the analog phone is connected to a VoIP system, and telephone calls are made using compressed voice (for example, G.729 at 8 Kbps), there is a substantial risk that the audio tones of the DTMF digits sent through the compressed voice path may become too distorted to be predictably recognized correctly by a remote voice mail system.

Even when you use uncompressed G.711 A-law/μ-law 64 Kbps for VoIP calls, there is still a risk that DTMF digits can get distorted in transit. This is because of the risk of packet loss in the VoIP network. If the network drops an occasional IP packet containing voice, this is usually imperceptible to the human ear. However, if an IP packet is dropped that contains an audio encoding of part of a DTMF tone, this is very likely to keep the DTMF digit detection in the far-end system from detecting the digit (or to make it erroneously detect multiple digits). This is also the reason that other nonvoice audio signals such as fax and data modems need special treatment in VoIP networks.

To work around this issue, you generally have to use some form of DTMF relay. DTMF relay causes the digit press to be detected by the PSTN trunk or analog phone interface on the VoIP gateway. The originating VoIP gateway then signals the digit as an explicit event to the far-end VoIP gateway and removes the audio signal for the DTMF from the voice packet stream. When the far-end VoIP gateway receives the signal for the DTMF event, it regenerates the DTMF audio signal and inserts it into the outgoing audio stream to the PSTN or analog phone.

In the case of the Cisco SCCP IP phones, the digit never exists as an audio signal from the VoIP perspective, because it's directly signaled via the SCCP control protocol. The digit audio that the phone user hears from the phone handset is for the benefit of the phone user only and is not passed to the VoIP connection.

Call Transfer and Call Forwarding in an H.323 Network Using H.450 Services

The ITU-T H.450 services are a set of standard supplementary services defined for H.323 VoIP networks. H.450 provides services above and beyond basic A to B telephone calls. Cisco CME 3.0 offers only the services related to call transfer (H.450.2) and call forwarding (H.450.3). Cisco CME 3.1 also introduced support for the H.450.12 capabilities discovery protocol to ease interworking issues with non-Cisco CME H.323 systems.

Here's a full list of the H.450.x services, including the date when they became formal ratified ITU-T standards:

- H.450.1 (2/1998)A generic functional protocol that supports supplementary services in H.323 (supported by Cisco CME)
- H.450.2 (2/1998)A call transfer supplementary service for H.323
- H.450.3 (2/1998)A call diversion supplementary service for H.323 (supported by Cisco CME)
- H.450.4 (5/1999)A call hold supplementary service for H.323
- H.450.5 (5/1999)A call park and call pickup supplementary service for H.323
- H.450.6 (5/1999)A call waiting supplementary service for H.323
- H.450.7 (5/1999)An MWI supplementary service for H.323
- H.450.8 (2/2000)A name identification service
- H.450.9 (11/2000)A call completion supplementary service for H.323 (includes callback for a busy subscriber)
- H.450.10 (3/2001)A call offering supplementary service for H.323 (includes camp-on busy subscriber)
- H.450.11 (3/2001)A call intrusion supplementary service for H.323
- H.450.12 (7/2001)A common information additional network feature for H.323 (for H.450.x capabilities discovery, supported by Cisco CME 3.1 and later)

An important thing to note about the dates these standards became available is that many H.323 networks were brought into service before these standards were issued. This means that support of these standards within H.323 networks varies widely, which causes some challenges in deploying these services within multivendor H.323 networks. Even within the overall Cisco voice product line, support of these services is not yet widespread.

Integrating Cisco CME in a SIP Network

Much of what you've read about linking Cisco CME systems over WAN VoIP links for H.323 also applies to SIP, so a lot of the heavyweight detail that's been covered for H.323 is not repeated here. Instead, the following sections focus on some of the differences between SIP and H.323 implementations. This approach also helps you understand some of the issues associated with investment protection for your VoIP network and hopefully provides some reassurance about picking the "right" protocol for intersite calls. Cisco Systems is agnostic about protocols. In that spirit, you'll see how Cisco CME provides you with flexibility and safeguards against protocol dependencies.

A major point that should be made here is that IOS and Cisco CME's support of SIP is primarily for SIP trunking, or using SIP as a protocol to connect calls between peer Cisco CME systems over a WAN link. This is primarily a property inherited from the Cisco IOS Voice Infrastructure functionality that underlies Cisco CME. This is quite a different usage case than that of connecting SIP phones directly to Cisco CME.

However, you can host SIP phones directly on Cisco CME 3.0, because the same Cisco IOS 12.3(4)T code also independently includes the Survivable Remote Site Telephony for SIP (SIP-SRST) feature that provides a basic Registrar and Redirect Server. The services and features that you can access from the SIP phones are very limited in comparison to the phone features offered for SCCP-based phones. More significantly, Cisco CME 3.x does not provide any mechanisms to support administration and configuration management for SIP phones.

You can also host H.323-based phones on a Cisco CME system if you use a router image that includes gatekeeper functionality. These services that enable support of H.323 and SIP phones are part of the general IOS Voice Infrastructure functionality and are unrelated to Cisco CME.

You should understand here that although you can concurrently and independently operate the IOS Voice SIP and H.323 phone-hosting capabilities with Cisco CME, this functionality is not integrated and productized in the same way as support for SCCP phones. Cisco CME 3.0, 3.1, and 3.2 are not marketed as providing SIP or H.323 phone support for this reason.

The following sections describe using SIP to interconnect Cisco CME systems.

Two-Node Topology with SIP

You can connect two Cisco CME systems using a pair of VoIP dial peers configured symmetrically on each CME to point to the other CME. This is exactly the same as the H.323 case described in the section "A Simple Two-Node Topology with H.323" near the beginning of this chapter. The only difference is that you must explicitly select the SIP protocol in your dial peer, whereas H.323 is the default VoIP protocol.

For a pair of Cisco CME systems that have extensions 1000 to 1099 on Cisco CME 1 (IP address 10.1.1.1) and 2000 to 2099 on Cisco CME 2 (IP address 10.1.2.1), you need the dial peers shown in Example 7-15.

Example 7-15. SIP Dial Peers

```
Cisco CME 1:
Router#show running-config
dial-peer voice 2000 voip
     destination-pattern 20..
     session target 10.1.2.1
     session protocol sipv2
     dtmf-relay sip-notify
     codec g729r8

Cisco CME 2:
Router#show running-config
dial-peer voice 1000 voip
     destination-pattern 10..
     session target 10.1.1.1
     session protocol sipv2
```

Summary

This chapter covered many details of the H.323 and SIP protocols. Choosing between the H.323 and SIP protocols depends on where you are starting from and what you are trying to do. In many cases, your choice is dictated by the nature of your existing voice network. This tends to favor H.323 simply because it is a more mature and well-established protocol. This is certainly the more conservative approach that you may favor if you're looking at immediate deployment of a large-scale VoIP network, and you don't want to get caught up in new technology issues.

If you are just getting started with VoIP trials, if you have no legacy network to worry about, and you are interested in exploring the latest new ideas in VoIP, SIP is a good choice.

In the end, your decision is best based on looking at solid business functions and economic factors. Your VoIP system end users are not likely to care what the underlying protocol is. You should be looking primarily to see which vendors have the best products and user features that are available within your deployment schedule.

In the end, the choice between H.323 and SIP is generally not an irrevocable decision. They are both very good protocols for implementing peer-to-peer voice networks. Cisco CME supports both protocols and will continue to evolve as they do. Many other vendors of VoIP systems are likely to follow a similar dual-track approach to this issue. It's unlikely that any radically new voice telephony features will be available using just one basic protocol. The key innovation is to use VoIP and embrace the concept of a converged voice and data network. You can do this with either H.323 or SIP.

Chapter 8. Integrating Cisco CME with Cisco CallManager

This chapter covers the following topics:

- Basic call interoperability between Cisco CallManager Express (CME) and Cisco CallManager
- Call transfer and forward operation between Cisco CME and Cisco CallManager
- Operation of name display between Cisco CME and Cisco CallManager
- H.450 tandem gateway operation

This chapter covers the deployment of Cisco CallManager Express (CME) for branch offices in conjunction with a Cisco CallManager deployed at a central or head office site. In this situation, the central Cisco CallManager site can act as a hub linking the remote Cisco CME sites. Cisco CME and Cisco CallManager can communicate across IP WAN links using H.323 (or using Session Initiation Protocol [SIP] with Cisco CallManager 4.1 or later versions).

In H.323 networks, Cisco CME provides supplementary service interworking (H.450) using voice over IP (VoIP) hairpin call routing when needed for intersite call transfer and forwarding. The basics of this operation are covered in Chapter 5, "Cisco CME Call Processing Features." Chapter 7, "Connecting Multiple Cisco CMEs with VoIP," covers inter-Cisco CME H.323 network operation. This chapter discusses these services in a network that has both Cisco CME systems and one or more Cisco CallManagers.

Goals of Interoperability

Chapter 2, "Building a Cisco IPC Express Network," covers the choice between centralized architectures (those based solely on Cisco CallManager) and distributed or autonomous site architectures (those based on Cisco CME) for implementing VoIP networks. In many real-world cases, this isn't a simple either/or decision, because many networks include both types of structures, discussed in Chapter 2 as the hybrid network architecture.

Real enterprise VoIP networks that have been designed consistently from the ground up and that adhere to a single uniform architectural approach are rare. The technologies available to network designers have evolved rapidly over the past decade or two. This rapid evolution is likely to continue for some time. It requires organizations to continually rethink their network architectures to take advantage of the latest available enhancements. Not only do the technologies change, but so do the companies trying to make best use of them. Companies split and merge and reinvent themselves in a continuous effort to stay profitable and competitive. This leads to real-world networks made up of a mixture of architectures formed by the ad hoc fusion of components contributed by multiple network designs.

Looking at VoIP networks that incorporate Cisco components, you commonly see both central-site Cisco CallManager networks using Survivable Remote Site Telephony (SRST) at some remote branch offices coupled with Cisco CME systems used at other remote offices. Being able to interconnect these systems is a fairly important consideration. In fact, some businesses deliberately design their networks using both central and distributed models to take into account issues with geographic variation in the availability of WAN services. For example, in the banking industry, central Cisco CallManager designs have been widely used in city branches located in metropolitan areas where adequate bandwidth and quality of service (QoS)-enabled WAN links are fairly readily available. On the other hand, Cisco CME systems have been used in small-town bank branches located in more rural areas where WAN services might be less sophisticated and unable to support voice.

Both Cisco CallManager and Cisco CME support H.323, which you can use to create Cisco CallManager-to-CME links. Cisco CME also supports SIP for VoIP interconnect. SIP is also being introduced as a WAN trunking interface on Cisco CallManager. This chapter focuses only on the H.323 interconnect option, because the SIP interconnect option is still a work in progress as SIP support on successive Cisco CallManager releases evolves. However, you can expect that most of the architectural issues raised in this chapter are also applicable in the SIP context.

The descriptions contained in this chapter apply to the Cisco CME 3.1 and 3.2 releases and the Cisco CallManager 3.3(3) and 4.0 releases. Newer releases may have different behaviors and options than those described here.

Basic Calls Between Cisco CallManager and Cisco CME

Even before the introduction of Cisco CME, Cisco CallManager used Cisco IOS voice routers to provide a Public Switched Telephone Network (PSTN) access gateway for Cisco CallManager's IP phones. Both the H.323 and Media Gateway Control Protocol (MGCP) VoIP protocols can support this function. The choice between these two is partly a historic issue and partly related to the type of PSTN interface used, but this topic is outside the scope of this book.

Direct MGCP integration between Cisco CME's IP phones and Cisco CallManager is not supported. Although this does not preclude the concurrent operation of MGCP (used for the PSTN gateway ports) and H.323-with-CME on the same Cisco IOS voice router, this configuration is not recommended. Using MGCP with Cisco CallManager to control the PSTN voice ports on a Cisco CME system would force PSTN traffic originated by the Cisco CME's IP phones to route through the Cisco CallManager to reach the PSTN. In most cases, this would be inefficient because it would result in the Cisco CME-to-PSTN voice traffic traversing the Cisco CME-to-Cisco CallManager WAN link twice.

Because Cisco CME is built on top of the Cisco IOS voice infrastructure software foundation also used by the Cisco IOS router-based PSTN gateways, Cisco CME inherits most of the H.323 gateway-to-CallManager interoperability. You should be aware of some minor differences, however.

In the simple PSTN gateway case, most of the call progress signaling for calls is performed as in-band audio tones. For example, when an IP phone (hosted by Cisco CallManager) places an outbound call to the PSTN, the ringing tone (also called the alerting or ringback tone) heard by the caller is usually generated as an audio signal that's passed through end-to-end from the PSTN trunk. This means that the audio path from the gateway to the IP phone is opened and active before the call is connected.

In the case where a call is placed from a Cisco CallManager IP phone to a Cisco CME IP phone, the ringing tone is provided as an out-of-band H.323 indication from Cisco CME (through H.323 control channel signaling). This means that the Cisco CME system signals to the Cisco CallManager, which in turn instructs the Cisco CallManager's IP phone to generate the ringing tone locally.

The reasons for this difference include the following:

- It saves some bandwidth because the audio path is not opened until the call actually connects.
- The Skinny Client Control Protocol (SCCP)-based IP phones attached to Cisco CME cannot generate in-band ringing tone.
- It avoids issues with shared extension lines, where an inbound H.323 call to Cisco CME may ring multiple phones at the same time. This could create complexity in choosing which phone should physically be required to do the tone generation. Add to this the fact that any of the Cisco CME phones involved might also have calls already in progress, and it should become clear why out-of-band signaling of call progress tones is the preferred approach for Cisco CME.

This issue of ringing-tone signaling causes some specific changes in the H.323 protocol exchange that Cisco CME uses when talking to a Cisco CallManager, compared with H.323 signaling to another Cisco IOS PSTN gateway, for example.

To get Cisco CallManager to provide local ringing-tone generation for outbound calls from Cisco CallManager, the Cisco CME delays negotiation of the media path until the call connects. The Cisco CallManager 3.3 and 4.0 H.323 implementations assume that in-band ringing tone is provided (by the Cisco CME or another H.323 device) on all calls that negotiate the media path (using H.245) before the call is connected. This delayed negotiation can lead to a minor delay (typically about a quarter of a second) in establishing the audio path when the call actually connects. This

Call Transfer

Cisco CallManager and Cisco CME implement significantly different approaches to handling call transfer. These differences are related to the basic architectural differences that exist between a highly centralized (CallManager) and fully distributed (Cisco CME) VoIP network architecture.

In Cisco CME, the preferred mechanism for handling call transfer is H.450.2. This allows calls to be transferred in a highly optimized fashion not only between phones on the same Cisco CME system, but also between different Cisco CME systems. This is a significant attribute when you consider that Cisco CME-based VoIP networks can include hundreds or thousands of individual CME nodes. Each CME node is a distinct and separate H.323 device with autonomous call processing.

In Cisco CallManager, the call transfer mechanism is designed to allow calls to be transferred between hundreds or thousands of IP phones controlled by the same Cisco CallManager (or CallManager cluster). Furthermore, in a Cisco CallManager environment, there is a significant need to separate the H.323 control path from the H.323 media path. Because a single Cisco CallManager server can be required to control approximately 2500 IP phones, it's impossible for the server to play an active intermediary role in the media path for all phone calls. Consider that there's a media packet in each direction every 20 milliseconds (ms) for each call, and then multiply this by 2500 phones. To allow a Cisco CallManager to support this number of phones, the media path for phone-to-phone calls must be directly between the phones whenever possible.

Note that each Cisco CallManager (or CallManager cluster) represents a single H.323 device from the external VoIP network perspective regardless of how many IP phones it supports.

One other issue to examine when comparing Cisco CallManager to Cisco operation CME is that for enterprise telephone systems, the ratio between internal and external call counts is related to the overall size of the phone system. As the number of extensions attached to a phone system increases, so does the relative proportion of internal calls compared to external calls. For example, in a system with only ten phones, almost 100% of phone calls are external calls between a phone and the outside world. In a system with 1000 phones, the external calls may make up only about 10% of the total call volume.

You can view this call transfer difference between Cisco CallManager and Cisco CME as being equivalent to the difference between an internal intra-private branch exchange (PBX) call transfer and an inter-PBX transfer. Viewed from the legacy PBX perspective, you can see that these are fundamentally different problems.

H.323 Call Transfer Using an Empty Capabilities Set

The problem with call transfer becomes more complex when you consider the interaction of Cisco CallManager IP phones with an external H.323 network. When an H.323 call exists between an external H.323 device and a Cisco CallManager IP phone, the preferred arrangement is to have a direct media path between the endpoints that does not pass through the Cisco CallManager server. In some cases, a direct media path is not possible (as described in the section "Call Transfer and the Media Termination Point"). In this case, it is necessary to introduce a Media Termination Point (MTP) into the media path. The MTP acts as a media relay or middleman and relays the Real-Time Transport Protocol (RTP) voice packets between the two terminating endpoints. The call signaling path does pass through the Cisco CallManager server. The H.323 signaling is terminated on the Cisco CallManager server and then is converted into SCCP to talk to the IP phone. The Cisco CallManager is required to participate in this signaling path to provide the needed conversion between H.323 and SCCP.

When there is a call transfer for an H.323 call from one Cisco CallManager IP phone (phone A) to another (phone B), the H.323 signaling path does not change. It remains terminated on the Cisco CallManager server. The Cisco CallManager server establishes a new SCCP signaling path to phone B. Of course, the media path also has to change. Changing the media path on the IP phones is easy. The Cisco CallManager simply sends the appropriate SCCP messages to phone B, telling it to participate in the media connection to the external H.323 endpoint.

To change the media connection on the H.323 side, the Cisco CallManager uses a mechanism known as Empty Capabilities Set (ECS). This mechanism informs the external H.323 device that it should stop sending its media

Call Forwarding

Call forwarding (for busy, no-answer, and unconditional forwarding) raises many of the same issues as call transfer. Likewise, these issues can be addressed using an MTP to simplify the H.323 signaling operations.

As you read in Chapter 7, the preferred method of handling call forwarding for Cisco CME is H.450.3. Again, Cisco CME can disable its H.450.3 feature when it detects calls from a Cisco CallManager. Under these circumstances, the Cisco CME system falls back to using internal call forwarding or VoIP-to-VoIP call routing for intersite call forwarding.

Just like the call-transfer case, an MTP allows internal call forwarding to occur without impact to the H.323 call leg. Of the three types of forwardingbusy, no-answer, and unconditionalthe no-answer form generally involves more signaling complexity. When a call forward no-answer occurs, the preliminary call negotiation for the original called phone must be revoked (after the no-answer timeout). It is replaced with a new call to the forward-to destination phone. The forwarded call can potentially require the use of different parameters than those negotiated for the original called phone. The busy and unconditional forms of call forwarding usually don't involve a preliminary call actually reaching the forwarding phone; this tends to simplify the signaling. Figure 8-6 shows call forwarding between a Cisco CME system and a Cisco CallManager using MTP.

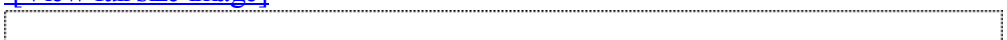
Figure 8-6. Cisco CallManager and Cisco CME Call Forwarding with MTP

[View full size image]

You can see that Figure 8-4, which shows call transfer, and Figure 8-6, which shows call forwarding, are nearly identical. Figure 8-7 shows intersite Cisco CME call forwarding for a call from a Cisco CallManager. Compare it to Figure 8-5 for the equivalent call transfer case.

Figure 8-7. Cisco CallManager Forwarding with Multiple Cisco CMEs

[View full size image]



Connected Party Name and Number Services

One of the reasons why Cisco CME uses the H.450.2 call transfer and H.450.3 call forwarding mechanisms is that it provides a standard means to support updating of the originating phone's display to track who the connected-to party is. Connected party display is often considered an important feature in business telephone networks. The connected party display is broader than the usual residential caller ID display you may be familiar with at home. Not only does it provide information about the calling party for inbound calls, but it also provides information about the called party for outbound calls. The connected party display can show you the name of the person you have called, for example. This is in contrast to simply displaying the number you dialed. Also, the standard caller ID display function provides a one-time indication of who the caller is only at the beginning of an incoming call. The connected party display feature allows for midcall updates of the connected party information that occur when the far end of the call performs a call transfer.

For example, after a caller has been transferred, his phone display can be updated to show the name and number of the extension he has been transferred to. Likewise, when you receive a call transferred to you by someone else, your phone display may initially show the caller ID of the person who is transferring the call to you. After the transfer is complete, your phone's display may update to show the caller ID of the transferee (caller).

When Cisco CME disables the H.450.2/3 mechanisms to interwork with Cisco CallManager, this display update mechanism is unavailable. However, Cisco CallManager has its own mechanism for performing connected party display updates. Cisco CallManager uses H.323 display and information IEs. Cisco CME also supports these IEs for performing connected party display updates.

The H.323 IE messages are carried as part of the H.323 signaling path. They are unaffected by the use of an MTP because they are informational messages only and don't generate changes to the call signaling state. This means that even when a Cisco CallManager performs an internal call transfer hidden behind an MTP, the H.323 information and display IEs are received by the Cisco CME system and are used to provide display updates. Not only can Cisco CME receive display IEs from Cisco CallManager, but it also sends display IEs to Cisco CallManager when Cisco CME performs internal transfer or forwarding.

Cisco CME sends these display IE messages for all H.323 calls regardless of whether a Cisco CallManager is involved in a call. This means that you can still get intersite connected-to party updates in Cisco CME networks where you have chosen to globally disable H.450 services.

Using H.450.x Tandem IP-to-IP Gateway

One final point to understand in planning your Cisco CallManager-to-Cisco CME connections is the advantages that an H.450 Tandem IP-to-IP gateway can provide. You can insert an H.450 Tandem IP-to-IP gateway into the call path between your Cisco CME network and Cisco CallManager, and use it to mitigate some of the issues that arise from the use of VoIP-to-VoIP call paths. If an intersite call transfer or forward initiated by a Cisco CME creates a VoIP hairpin call, you can often use an H.450 Tandem IP-to-IP gateway co-located with your Cisco CallManager to avoid most of the voice path delay caused. Just hairpin the media stream through the Cisco CME system located at the end of a narrow-bandwidth WAN link. Refer to Chapter 7 for more details about IP-to-IP gateways.

Summary

In this chapter, you've learned how to connect a network of Cisco CME systems with a Cisco CallManager using H.323. You've seen how Cisco CME can automatically and intelligently react to the presence of a Cisco CallManager and adapt its behavior accordingly. You've also learned how to optimize call transfers and forwards using H.450 for intersite Cisco CME calls within a fully distributed telephony network that at the same time can interoperate with a central site Cisco CallManager.

Chapter 9. Cisco IPC Express Automated Attendant Options

This chapter covers the following topics:

- Using an automated attendant (AA) or receptionist in your office
- Understanding the Cisco IP Communications (IPC) Express integrated AA
- The Cisco Unity Express (UE) system AA, including features such as dial-by-number, dial-by-name, business hours, and holiday call routing
- Customizing the Cisco UE AA with prompts and scripts using the Cisco UE AA Editor
- The Cisco UE Greeting Management System (GMS)
- Tasks to set up a Cisco UE AA
- Toolkit Command Language (TCL)-based AA as an alternative to the Cisco UE integrated AA

The chapters thus far in Part II, "Feature Operation and Applications," have discussed how to use the Cisco CallManager Express (CME) call processing and IP phone features and operation. Cisco CME provides Cisco IOS router-based call management and control to the IP phones and public switched telephone network (PSTN) trunks defined on the system. An IP telephony package for a small or medium office or enterprise branch office is, for most installations, incomplete without automated attendant (AA) and voice mail functions, even though not everybody needs these features.

An AA is a voice system feature that automatically answers your business's incoming calls with a welcome greeting (for example, "Thank you for calling company x."). It also provides a menu of choices for the caller to select from. The caller presses the appropriate button on the phone. The AA system then interprets the dual-tone multifrequency (DTMF) tones and redirects the call to the chosen destination based on the menu selection.

This chapter discusses the Cisco IP Communications (IPC) Express options for AA functionality, including an integrated AA application and a Toolkit Command Language (TCL)-based AA. The integrated AA option is a Cisco CME application, Cisco Unity Express (Cisco UE), which includes both an AA and a voice mail component. The AA component is covered in this chapter, and the voice mail portion of the application is discussed in Chapter 10, "Cisco IPC Express Integrated Voice Mail."

The Cisco UE AA is a full-featured, customizable AA application offering an easy-to-use graphical user interface (GUI) drag-and-drop interface to change the system's AA menu flow. Cisco UE is cost-effective for small and medium standalone or networked offices, or as a distributed AA (and voice mail system) in branch offices of an enterprise network. Cisco UE offers up to five distinct AAs, each associated with an individual pilot number and one or more scripts that can be fully customized.

The second alternative is a TCL-based AA that leverages Cisco IOS built-in TCL capability. Although it can be customized by someone proficient in the TCL language, it is not typically an end-user interface. Instead, it is a tool used by Cisco and Cisco developers to provide ready-made scripts for your system.

Using an Automated Attendant or a Receptionist in Your Office

With its AA options, Cisco CME can provide a fully automated front end to calls coming into a small office. All IP phone extensions, the system's voice mail pilot number (the number called to access the voice mail application), and general information such as location, business hours, and driving directions can all be accessed via the AA menu choices. However, many small businesses choose to enhance their customer relationships with a receptionist instead of presenting an AA menu to their callers. For maximum flexibility, both modes of operation can be used. A receptionist can answer calls during normal business hours, and an AA can direct after-hours calls or can be used as a backup if the receptionist is unable to answer the phone.

When deploying an AA in an office, you must first decide how to handle calls into the office, both during business hours and after hours. Next, craft the Cisco CME call routing features, the AA menus, and the digit manipulation done by the Cisco CME system to achieve the desired call routing. Cisco CME call routing capabilities depend in part on how the PSTN trunks are configured and what digit information the PSTN delivers.

Calls to the Main Office Number

For a business with direct inward dial (DID) service, the only calls typically answered by an AA or receptionist are the general calls to the main office number. DID calls are switched automatically to the extension number that the caller dials. For example, extension 3001's PSTN DID number is 222-555-3001, so whenever this number is called, the IP phone for extension 3001 rings automatically without AA or human assistance. DID call routing makes sense for a business such as a lawyers' office, where each lawyer must be individually reached by clients and the DID PSTN phone number appears on the lawyer's business card.

Cisco CME's default operation is to switch calls automatically to the dialed extension (via dial peer matching), provided that dialed digits (DNIS) are available from the PSTN for the call. However, even with DID service, it is still possible to redirect all or certain calls to an AA or receptionist by doing digit manipulation on the dialed digits if you prefer this call routing.

Many small businesses might not desire the cost of DID PSTN service because of either size (the business is too small) or the type of business they conduct. For example, a small charity or restaurant's incoming phone business is of a general nature and can be handled equally well by any employee answering the phone. For these types of businesses, multiple general PSTN lines make more sense.

Calls to multiple general PSTN lines can also be directed to either an AA or receptionist. Or all calls can simply ring on the multiple line appearances (Line 1, Line 2, and so on) on the phones and be answered by any employee at any phone. It is also possible that these PSTN lines appear on one phone onlythe receptionist's phoneand that the receptionist then transfers the call to the appropriate extension. All calls not answered directly by employees via these methods must be answered by either a receptionist or an AA.

If you decide that only an AA is required in an office, simply configure all main office PSTN calls to terminate on the AA pilot number. As soon as the AA answers the call, callers can use the dial-by-number or dial-by-name features to reach the extension or service of their choice.

If you decide that a receptionist is required, configure all main office PSTN calls to terminate on the receptionist's extension. The receptionist can then transfer the call to the extension of the employee or group the caller wants to contact. If you want to use the AA to handle calls as a backup to the receptionist, or as an after-hours mechanism, it is possible to call-forward-no-answer (CFNA) the receptionist's extension to the AA.

Note

If the receptionist's phone CFNAs to the AA, unanswered calls cannot be directed to a personal voice mailbox, with the result that the receptionist cannot have voice mail. This situation can be circumvented in one of two ways:

- Define a separate private extension for the receptionist that forwards to personal voice mail when not

Understanding the Cisco IPC Express Integrated Automated Attendant

The preceding section discussed general considerations with call routing to a receptionist or an AA. Provided that your business requires at least some of its calls to be routed via an AA, the following several sections discuss Cisco UE's AA capabilities in more detail. The features discussed here assume a minimum of Cisco UE 1.1 software. Some features require later releases.

When callers call into the AA number, based on the time of day and the day of the week, the AA may play a different greeting to callers, followed by a menu of options. After the caller selects the desired option, the call is routed to the requested destination, which can be an employee extension or a service such as recorded driving directions or business hours.

One of the many benefits of an AA is that after-hours callers can still receive the information they need to contact your business. It also lowers business expenses and lessens the burden on a receptionist. An AA is especially useful for larger sites where the number of calls to the main office number may overwhelm a receptionist. An AA menu should be carefully designed, because it is the public face of your business to clients. Badly designed menus, menu choices that are ambiguous, or loops within the AA script will frustrate callers.

Cisco UE provides the capability to define up to five different AAs. Each individual AA is a menu script associated with a phone number called a pilot or call-in number. Calls arriving at the pilot number, either internal or PSTN calls, are then handled by the script associated with the pilot number.

Cisco UE ships with a system AA (using system-provided scripts) that requires minimal configuration to set up for your business. You may also use one or more custom scripts that tailor the AA menu options to your business's exact needs. Each pilot number that you associate with an AA script counts as a distinct AA application. Up to five of these AA applications can be active in the Cisco UE system at once.

Which AA is executed depends on which pilot number you direct your calls to; this is entirely flexible. For example, you may have the system AA active, one custom AA that you have written, and one or more test scripts that are not actively taking calls but that are present on Cisco UE as the third or fourth AAs. Or you may have different departments, each with its own direct PSTN number, and each wanting to manage its own AA greeting and menu separate from the other departments. This configuration can be accommodated with two or more separate AAs in the system. Most businesses, however, in addition to an optional receptionist use one of the following:

- The system AA
- A single custom AA (with a dormant system AA, meaning that the system AA is present but no calls are directed to it)

The Cisco UE System AA

The system AA is shipped to ensure that you can set up a working system in a minimal amount of time. You simply configure a small number of parameters, and record a few greetings (called prompts on the Cisco UE system). You do not have to worry about the script itself, because it is fixed and cannot be changed.

The system AA offers the canned menu shown in Example 9-2. The highlighted lines are new features added to the system AA in Cisco UE Release 2.1.

Example 9-20. Cisco UE System AA Menu

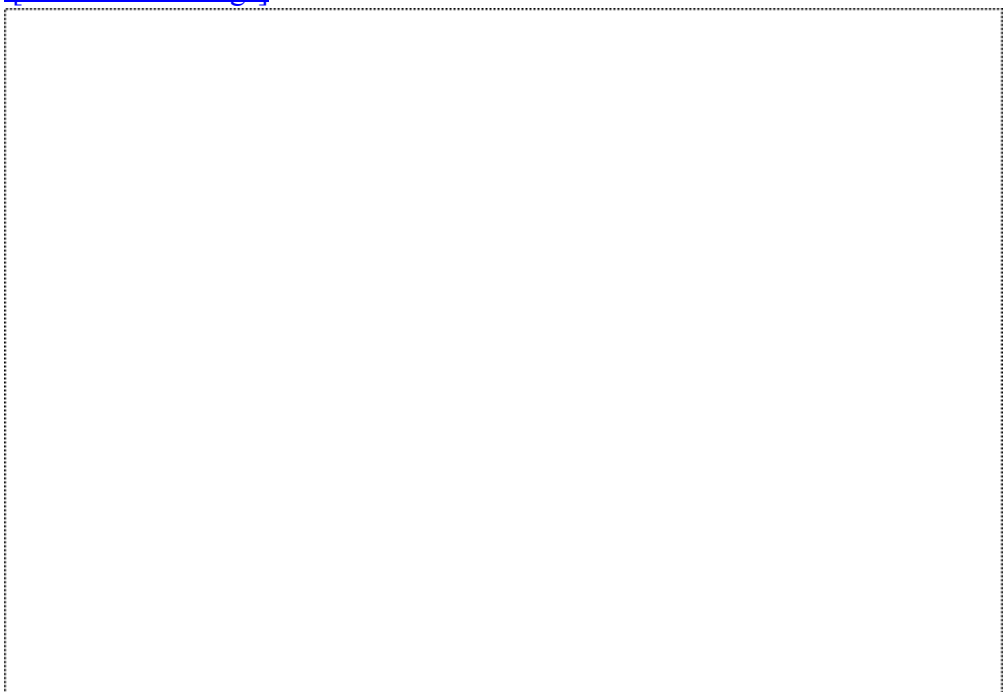
```
If emergency alternate greeting active
   Play "alternate greeting"
Play "welcome greeting"
If holiday
  Play "holiday greeting"
Else if business open
  Play "business open greeting"
Else
  Play "business closed greeting"
To enter the phone number of the person you are trying to reach, press 1
To enter the name of the person you are trying to reach, press 2
To transfer to the operator, press 0
```

You can change the welcome greeting to your own greeting, such as "Welcome to XYZ Company...." Similarly, you can rerecord the alternate, holiday, business open, and business closed greetings to suit your needs even though the Cisco UE system supplies system default greetings.

The system AA's menu structure cannot be changed. The parameters of the system AA script, such as the welcome greeting and the operator extension, can be configured as shown in Figure 9-2. The list of available prompts (or greetings) present on the system is shown in the drop-down box. The operator extension is set to 3010 in Figure 9-2. The concepts of scripts, variables, and parameters are discussed in more detail in the "Customizing the Cisco UE AA" section later in this chapter.

Figure 9-2. Setting the System AA Parameters

[View full size image]



Customizing the Cisco UE AA

The system AA is fairly simple, and the menu flow cannot be customized. (The only customization possible is rerecording the greetings, setting the business hours and holiday dates, and setting the operator extension.) If a different AA menu flow is needed, a custom AA script must be placed on the system. Up to five AAs (including the system AA) can be loaded into the Cisco UE system to provide a wide range of flexible AA menus to cover very specific business needs.

This section of the chapter covers how to prepare a custom AA script using the Cisco UE AA Script Editor. You'll explore script elements such as steps, parameters, and variables. You'll learn about operations such as viewing, uploading, downloading, and executing scripts. System limits governing scripts and prompts on your system are also discussed.

AA Scripts, Prompts, Variables, and Parameters

To understand how to build a custom AA script, some programming knowledge is helpful. However, even if you do not have any programming background, customizing an AA script is not a complex task. The Cisco UE Script Editor provides a graphical drag-and-drop interface for easily rearranging the steps in a script. An AA script is essentially a computer program, which means that it is a list of steps executed for every call that arrives at the system. The sequence of these steps is specified at a very high level with an easy-to-use icon-based GUI interface.

An AA script is an ordered, logical sequence of actions called steps. Each step signifies a task or operation to be performed on the call, such as playing a prompt, collecting DTMF digits, or redirecting a call to another destination. Each script step has one or more attributes associated with it, such as the Menu step in which the attributes describe what should be done when the caller presses 1 or 2 or any other DTMF digit on the phone keypad.

The following elements make up an AA script:

- Steps Programming steps in the script perform actions such as answering the call, playing prompts, receiving DTMF digits from the caller, making conditional choices based on time-of-day or any other internal value, transferring a call, looking up a username in a directory, and many more.
- Prompts Scripts use short sections of recorded voice to interact with the caller, such as "Please enter the extension of the person you want to reach," "Press 1 for sales," or "Our business hours are 8 a.m. to 5 p.m. Monday through Friday." Each of these recorded voice segments is called a prompt and is referenced by the script as a variable. A prompt is physically stored in a .wav file on the system.
- Variables A variable is a label or tag representing a value that may change during the life of the script. For example, retry_limit may be a variable referenced in a script. This tag has a value of three or five, indicating how many times you want to retry a prompt after a timeout.
- Parameters A parameter is a variable whose value can be changed while the script is running on the Cisco UE system. In other words, you can change the value without editing the script and reuploading it to the system. For example, operExtn is a parameter that can be populated with extension 3001 on Monday and 3002 on Tuesday and so on without the need to edit the script. Not all script variables have to be parameters. Only those that you want to change without editing the script are assigned the property of parameter.

Preparing a Script Using the Cisco UE AA Editor

Custom AA scripts are executed on the Cisco UE system, but they are prepared offline on a Windows PC platform by using the Cisco UE AA Editor. Follow these steps, which are shown in Figure 9-4, to prepare and install a custom Cisco UE AA script:

The Cisco UE Greeting Management System

Cisco UE is a low-end entry-level system that does not assume that you have sophisticated offline recording facilities or a studio available to create your AA prompts. You can certainly use such facilities if you have them and upload the resulting professionally created prompts to the system. Cisco UE also offers a built-in facility to record and manage prompts via any telephone that can call into the system (a local IP phone or PSTN phone). This feature is called the GMS.

Like the AA and voice mail pilot numbers, the GMS also has a pilot number assigned. When calls arrive at the GMS pilot number, the caller is put into an interactive session where you can listen to existing prompts and record new ones.

Calling into the GMS requires an extension and personal identification number (PIN) for security authorization. The extension/PIN combination is the same one used for your mailbox access authorization. Any user defined on the Cisco UE system with a mailbox can, therefore, theoretically access the GMS. However, actual access is restricted to users who also have administrative privileges on the Cisco UE system (those who belong to the administrators group). This is much more likely to be just one or two of the users defined on the system.

The GMS itself runs a Workflow script within the Cisco UE system and can, therefore, be seen when a listing of all the system scripts is given (the script is called promptmgmt.aef). The GMS offers two main features related to the Cisco UE AA, as discussed in the following sections:

- Recording or deleting the EAG
- Recording and listening to custom prompts

Prompts on the Cisco UE system must be a specific .wav file format to be played correctly. You can record prompts with the GMS or offline with a PC or studio sound equipment and then upload them to Cisco UE. You can download prompts already resident on Cisco UE to your PC as .wav files. Uploading and downloading prompts works just like uploading and downloading scripts, as discussed earlier, in the section "Uploading and Downloading Scripts."

Recording or Deleting the EAG

You can record the EAG via the GMS. Calling into the GMS guides the administrator through the process of recording or deleting the alternate greeting:

Step 1. The script prompts the caller (administrator) to enter his or her extension and PIN to determine if the caller has administrative rights on the system.

Step 2. After the caller is authenticated, the script checks to see if an EAG exists on the system. If it does not, the script prompts the administrator to record one.

Step 3. As soon as the EAG is recorded and saved, it becomes active, and new callers to the AA start hearing the EAG before the regular welcome prompt.

If the EAG already exists on the system, the script prompts the administrator to delete the existing greeting. If the administrator deletes it, the EAG is deleted from the system, and new callers into the AA hear only the regular welcome greeting as their first prompt.

The system has no default EAG. If the EAG exists, it must have the filename AltGreeting.wav. If you record the EAG via the GMS, it automatically has this filename. If you record it on an offline system and upload it to the Cisco UE system, you must ensure manually that the .wav file has this name so that it will be recognized by the Cisco UE system as the EAG.

Setting Up a Cisco UE Automated Attendant

In the preceding sections, you learned the reasons to use an AA; general features of the Cisco UE AA, such as dial-by-number; and how to customize an AA script.

This section walks you through building the actual AA. Just having written the script is not enough. It is the biggest step, but now the script, the prompts, and the AA pilot number must be linked to form a working AA application on your Cisco UE system. This is not a difficult task, but it takes a few steps to accomplish:

Step 1. Determine whether the system AA is sufficient.

Step 2. Prepare a custom script.

Step 3. Upload the script.

Step 4. Record the prompts.

Step 5. Select a script.

Step 6. Set the script parameters.

Step 7. Assign a pilot number.

Determining Whether the System AA Is Sufficient

The first question to consider is whether you have to write a custom AA at all. Have a good look at the Cisco UE system AA, shown earlier in Example 9-2. Even though it's simple, it might be sufficient for your AA needs. If so, you only have to associate an AA pilot number with this script, customize the greeting, and insert your business hours and holidays, and you're done.

If the Cisco UE system AA is insufficient, you should continue using the sample AA functions illustrated in the various sample scripts in this chapter to put together a script that is tailor-made for your business.

Preparing a Custom Script

Follow these steps to prepare a custom script:

Step 1. Ensure that the Cisco UE Editor is installed on your PC. If it isn't, download it from Cisco.com and install it.

Step 2. Download the sample AA scripts from Cisco.com, and load them onto your PC.

Step 3. Draw a rough diagram of the flow of menus and prompts you want for your AA.

Step 4. Copy the sample script that is closest to your needs to a new file, and open it using the Cisco UE Editor.

Step 5. Adjust (edit) the script to match your drawing from Step 3 by copying and pasting steps from other sample scripts, rearranging the existing steps, or dragging new steps from the Editor's Palette Pane.

Step 6. Ensure that all the variables you defined and that you want to control via the Cisco UE GUI while the script is running are tagged with the parameter attribute. Also ensure that all the variables you want to control are defined in the top-level script and not in any subflows. Subflow parameters are inaccessible via the Cisco UE GUI to change their values.

TCL-Based Automated Attendant

As discussed at the beginning of this chapter, Cisco CME offers two options for an AA application:

- The integrated AA application, which is part of Cisco UE, as discussed in this chapter
- Leveraging the router's TCL scripting capabilities to provide an AA

Both schemes are adequate to provide AA functionality for your Cisco CME. However, when you're deciding which of these two mechanisms to use for your business, you should keep in mind several considerations, as listed in Table 9-6.

Table 9-6. Considerations When Choosing a Cisco CME AA

Consideration	TCL-Based AA	Cisco UE AA
Where the scripts execute	Scripts run on the router's processor and, therefore, have a small impact on the router's performance.	Scripts execute on the dedicated Cisco UE module. Therefore, they use the dedicated module CPU and do not affect the router's performance.
Memory	Scripts are loaded into the router's DRAM, so memory is allocated regardless of whether the script is being used.	Scripts are loaded into the dedicated Cisco UE module's storage and, therefore, do not use any router memory.
Scripting expertise	Requires a working knowledge of TCL programming and uses a text-based editor.	Uses an easy-to-use web-based graphical editor. Knowledge of logic and call processing steps is still helpful.
Error checking	The editor has no script validation or error checking.	The Cisco UE Editor has a validation function.
Script management	Requires an offline editor. Scripts are installed by copying the script to the router's Flash via Trivial File Transfer Protocol (TFTP).	Requires an offline editor. Scripts are installed by using a GUI Upload function that browses your PC file system.
Prompt management	Prompts are recorded offline as .au files and are copied to the router's Flash using TFTP.	Prompts can be recorded online using the Cisco UE GMS (from any phone). Prompts also can be recorded offline as .wav files and Uploaded via the GUI.
Hardware required	Router-based capability that does not require any extra hardware in addition to your Cisco CME router, although it may require an increase in	Cisco UE AA runs on the Cisco UE network module (NM) or advanced integration module (AIM) hardware blade that you slide into a slot on

Summary

In this chapter, you learned ways to provide an AA for a Cisco CME system. You have two options: using the AA that is a component of Cisco UE or using a TCL-based AA.

At the start of this chapter, you looked at general considerations when deploying an AA feature or using a receptionist for your business. You explored how to use both functions to offer personalized service to your clients and accessibility after hours or when the office is closed.

You learned about the Cisco UE AAhow to set it up, how to customize the scripts to suit your business needs, and how to work with the Cisco UE AA Editor. The end of the chapter provided information on the Cisco CME TCL AA. Trade-offs between using the Cisco UE AA and TCL AA were discussed.

Although Cisco UE is not the only means of offering AA with a Cisco CME system, it is by far the most integrated and comprehensive solution.

Chapter 10. Cisco IPC Express Integrated Voice Mail

This chapter covers the following topics:

- Cisco Unity Express (UE) voice mail application
- Subscriber voice mail features
- Caller voice mail features
- Administrative voice mail features
- Call redirection into voice mail
- Working with users and names
- Dialplan considerations
- Voice mail networking
- Voice mail deployment considerations

In Part I, "Cisco IP Communications Express Overview," you learned about Cisco Unity Express and its contribution to the Cisco IP Communications (IPC) Express solution. Cisco UE offers an automated attendant (AA) and a voice mail application integrated with the Cisco CallManager Express (CME) call processing software.

Chapter 9, "Cisco IPC Express Automated Attendant Options," covered the Cisco IPC Express AA options in detail, including the AA offered by Cisco UE. This chapter explores Cisco UE's voice mail capabilities in greater depth. Alternative voice mail options that can be used with Cisco CME are covered in Chapter 11, "Cisco CME External Voice Mail Options."

Cisco UE is a typical voice mail system that includes all the common features you would expect from a low-end voice mail system, including personal mailboxes; subscriber login control; sending, listening to, saving, deleting, and forwarding voice messages; recording greetings and a spoken name for each mailbox; and directing calls to voice mail after the user's phone has rung for a configurable number of cycles.

Additionally, Cisco UE offers voice mail networking between different sites in your network, broadcast messaging, and distribution lists. Cisco UE also provides a number of general delivery mailboxes (GDMs) that you can use for functions in your organization (such as the factory or the sales desk) as opposed to being assigned to an individual employee.

Cisco UE Voice Mail Overview

Cisco UE is a small-to-medium office or enterprise branch office voice mail system that you can use with either Cisco CME or Cisco CallManager as the call control agent for directing the calls and managing the IP phones. When deployed with Cisco CME, Cisco UE is a local, collocated voice mail system integrated into the same router chassis as Cisco CME. With Cisco CallManager, Cisco UE represents a distributed voice mail system with a centralized call control agent in the network. Because this book focuses on Cisco CME, this chapter covers only Cisco CME network deployments and operation in detail. If you are interested in how you can use Cisco UE voice mail in Cisco CallManager networks, consult Cisco.com for more information (http://www.cisco.com/go/cue). Cisco UE's voice mail capabilities are the same for both deployment choices.

Cisco UE is offered in two hardware form factors: a Network Module (NM-CUE) and an Advanced Integration Module (AIM-CUE). The Cisco UE hardware provides for a fully self-contained software and hardware system with an onboard CPU and operating system, memory, and storage capacity (a hard disk on the NM and compact Flash [CF] on the AIM). This relieves the router's processing of the tasks required to execute and manage AA menus and voice mail messaging. For this reason Cisco UE does not impact the router's performance at all.

The Cisco UE hardware draws power from the router chassis. All communication between the router and Cisco UE software is carried across the router's backplane. The Cisco UE module is inserted into the router, an IP address is assigned, and the system is active. This complete integration means that Cisco UE requires no external servers or cabling and only minimal configuration and setup to deploy as a fully functional voice mail system. Cisco UE provisioning and installation are covered in detail in Chapters 13 and 14.

Cisco UE requires a minimum of Cisco IOS software release 12.3.4T (Cisco CME 3.0) on the router, using the IP Plus or IP Voice minimum software image. The AIM-CUE was introduced later and requires Cisco IOS software release 12.3.7T (Cisco CME 3.1) or later. It is recommended that Cisco UE deployments use Cisco CME 3.2 (12.3.11T) or later.

The following sections further explain the following aspects of Cisco UE:

- The mailbox licensing structure and how application parameters are controlled by licensing
- The types of mailboxes offered by Cisco UE, including personal and GDMs
- The concepts of users and groups, which underlie Cisco UE mailbox definition and operation

Cisco UE Licensing

Cisco UE offers an entry-level voice mail system that is cost-effective for offices requiring up to 120 total mailboxes. You can purchase mailbox licenses at the 12, 25, 50, and 100 personal mailbox levels. Future enhancements in Cisco UE releases beyond 2.1 may offer higher mailbox levels. The mailbox license you purchase equals the number of personal mailboxes on the Cisco UE system. In addition, you get a small number of GDMs with the license (the difference between the 100-mailbox license and getting 120 total mailboxes on the system). GDMs are discussed in more detail in the next section. Table 10-1 summarizes the Cisco UE system parameters controlled by the system license for Cisco UE software releases up to 2.0.

Table 10-1. Cisco UE Voice Mail Licensed Parameters for Cisco UE Up to Release 2.0

Hardware Form Factor	Parameters	12-Mailbox License	25-Mailbox License	50-Mailbox License	100-Mailbox License
NM-CUE	Personal	12	25	50	100

Subscriber Features

A subscriber is the person who owns the voice mailbox. A subscriber can access the voice mailbox, listen to the messages in it, and take action on these messages using the Telephony User Interface (TUI). Cisco UE is not a unified messaging system and, therefore, does not allow access to voice messages by any means other than the TUI.

The following voice mail features are available to a subscriber:

- Mailbox login and PIN
- Mailbox tutorial
- Personal greeting
- Alternate greeting
- Spoken name
- Message management
- Message playback control
- Message waiting indicator
- Mailbox-full notification
- Message reply and forward
- Private or urgent messages
- Envelope information
- Message playout sequence
- Zero-out destination
- Local and remote nondelivery notifications
- Distribution lists

Caller Features

Two types of callers interact with a voice mail system:

- A customer or vendor who calls your business. The person he or she wants to speak to is unavailable or is already busy on the phone, and the call forwards to voice mail. The caller hears the greeting of the person he or she called and then leaves a voice message for that person.
- An employee of your business who calls into the voice mail system to check or retrieve his or her messages.

All callers interact with the voice mail system by using the TUI only. The following features are available to the first type of caller (the person calling to leave a message) and are covered in this section:

- Outbound greeting bypass
- Message editing
- Urgent messages
- Mailbox-full indication
- Message leaving and mailbox login
- Zero-out destination or revert to AA

The second type of caller is a subscriber. Features available to this type of caller were covered in the section "Subscriber Features."

Outbound Greeting Bypass

If callers are not interested in listening to the outgoing greeting of the mailbox reached, they can bypass the greeting and proceed immediately to the beep, where they can leave a message. The caller can bypass the outgoing greeting of a voice mailbox by pressing # on the phone keypad at any time during the greeting playout.

Message Editing

After recording a message, a caller can listen to, edit, rerecord, or delete the message before sending. Or he or she can simply hang up, and the message is automatically sent.

Urgent Messages

If a caller hangs up after leaving a voice message for a subscriber, the message is sent with normal priority. If a caller presses the # key to end recording of a message, the Cisco UE voice mail system provides a menu where the caller can choose to tag the message as urgent (among other options, such as rerecording the message, deleting it, or listening to the message before sending).

Urgent messages are played at higher priority in the recipient's mailbox so that a subscriber hears urgent messages

Administrator Features

As the system administrator, you can use the GUI or CLI to manage voice mail system parameters and voice mail subscriber accounts. The GUI is more user-friendly for individual operations, and the CLI lends itself better to scripting from another management system for faster configuration.

The voice mail-related features available to a Cisco UE system administrator include the following:

- Voice mail pilot number
- Mailbox management
- MWI
- Mailbox storage allocation
- Maximum message size
- Message expiry
- Subscriber PINs and passwords
- Broadcast messaging
- Voice mail operator
- Distribution lists
- Language support
- System reports and status

Chapter 14, "Configuring and Managing Cisco IPC Express Systems," covers the configuration steps of the Cisco UE voice mail system in greater detail. This section discusses the features themselves and how they operate.

Voice Mail Pilot Number

The voice mail pilot number is the number subscribers call to retrieve their voice messages (typically triggered by pressing the messages button on the phone), as well as the number IP phones are call forwarded to so that callers can leave a voice message.

Note

For Cisco UE, the voice mail pilot number is always different from the AA pilot number. The same pilot number

Call Redirection into Voice Mail

As mentioned in the section "Message Leaving and Mailbox Login," Cisco UE uses the last Redirected Number field in the call information to select the mailbox greeting to play to a caller. This field can be seen in the Diversion header in the SIP INVITE message delivering the call to Cisco UE, as shown in Example 10-5. In this example, 6800 is the voice mail pilot number, extension 7010 originates a call to extension 5010, and the call is redirected to voice mail by CFNA.

Example 10-5. SIP INVITE for a Call Forwarded to Cisco UE

```
router#debug ccsip messages
INVITE sip:6800@1.3.6.179:5060 SIP/2.0
Via: SIP/2.0/UDP  1.3.6.4:5060
From: "7010" <sip:7010@1.3.6.4>;tag=691AE6E4-223C
To: <sip:6800@1.3.6.179>
Call-ID: E5D39E6A-8FC011D7-9025DAEC-459632B0@1.3.6.4
CSeq: 101 INVITE
Max-Forwards: 6
Remote-Party-ID: <sip:7010@1.3.6.4>;party=calling;screen=no;privacy=off
Timestamp: 1054070868
Contact: <sip:7010@1.3.6.4:5060>
Diversion: <sip:5010@1.3.6.4>;reason=no-answer;counter=1
```

Correct voice mailbox selection depends on the redirection information delivered with the call, which is extension 5010 in Example 10-5. A call can be redirected to voice mail in numerous ways, and these ways may populate the redirection information differently. The following sections explore some common ways for calls to be redirected to voice mail and their dependencies on voice mailbox selection and operation.

Call Forward into Voice Mail

CFNA, CFB, and CFA are the typical ways to divert a call to voice mail. If a call is redirected multiple times before reaching voice mail, the value in the last Redirected Number field always selects the voice mail greeting that Cisco UE plays to the caller. For example, if extension 3001 calls extension 3061, which CFAs to extension 5001, which in turn CFNAs to voice mail, the caller at extension 3001 hears the voice mail greeting of the subscriber associated with extension 5001.

Some voice mail systems use the original called Number Field for voice mailbox selection. In the call flow just described, that would be extension 3061. This field does not change, regardless of how many times the call is subsequently diverted before reaching voice mail. Some voice mail systems, including Cisco UE, use the last Redirected Number field. Other systems allow you to configure which field to use.

Transfer and Conference

Other possible ways for a call to enter a voice mail system are by using call modification features such as transfer and conference. Call transfer to voice mail is fairly common, but call conference is perhaps less so.

If your business has a receptionist or administrative assistant answering calls, a caller may choose to transfer to the employee's voice mail instead of waiting to speak to the person. This scenario is discussed in detail in Chapter 9 in the section "Transferred Calls That Forward to Voice Mail."

When a call is transferred to voice mail, the extension initiating the transfer is the last redirected number, and this extension's mailbox is selected. For example, suppose a PSTN caller calls the receptionist at extension 3001. The caller wants to speak to User5 at extension 3005. If the receptionist transfers the call to voice mail, extension 3001's mailbox is selected. Chapter 9 describes a workaround configuration that allows your receptionist to transfer a call without ringing the destination extension (3005) while having the caller hear extension 3005's voice mail greeting.

Working with Users and Names

The Cisco CME and Cisco UE configurations include various username fields. Understanding which Name field controls which aspect of your system helps you configure your Cisco IPC Express system, including voice mail, to the best benefit for your business.

Names are associated primarily with users, but also, in some specific cases, with phones. Cisco CME and Cisco UE mutually identify a user by a unique user ID field. Before discussing the various Name fields in more detail, it is necessary to understand how a Cisco IPC Express system handles user IDs.

User ID

A user ID is a tag the system uses to track and identify a user and his or her associated configuration. Cisco CME and Cisco UE use the user ID tag to coordinate configuration information about a specific user between the router's configuration (Cisco CME) and the configuration information stored on Cisco UE's local disk or Flash unit.

The user ID is a single configuration field, but it shows up in various locations in a Cisco IPC Express system configuration:

- The Configure > Phones screen as the Login Name field
- The Configure > Users screen as the User ID field
- The Voice Mail > Mailboxes screen as the Mailbox Owner (User/Group ID) field
- The ephone Username field in the router CLI

This user ID uniquely identifies each user. It cannot change after the user is defined. If it must change, the user must be deleted and reinserted into the system's configuration. You want to avoid doing this, so plan user ID assignments carefully before entering them into the system configuration.

Caution

Deleting a user also deletes his or her associated voice mailbox. The action of changing a user ID, therefore, amounts to redefining the user's mailbox and losing (deleting) all of the user's current voice mailbox contents. Clearly this is undesirable, which is why user IDs typically never change after the system is configured and your employees start using it for their daily tasks.

If a user preexists in the Cisco CME configuration before Cisco UE is installed and configured, this user ID is automatically imported from the Cisco CME configuration into the Cisco UE Initialization Wizard. (This wizard is further discussed in Chapter 13.)

Note

If the Cisco CME CLI ephone-dn username configuration is changed by using the router's CLI, and the Cisco UE GUI is then synchronized with the latest changes, a new user is created and shows up in the configuration. The mailbox associated with the old user definition cannot be moved to the new user.

Name Fields

Dial Plan Considerations

Cisco UE deployed with Cisco CME requires that the IP phones that have mailboxes all have extensions of the same length. The specific length of the extension does not matter: the length of the extensions can be anywhere between one and 16 digits (as supported by Cisco CME). What is important is that all extensions with mailboxes must be of the same length on a particular Cisco CME and Cisco UE system.

Chapter 7 covered various ways in which you can manipulate digits on dialed numbers to translate PSTN numbers (E.164 numbers) into local extensions:

- Cisco IOS dial peer commands
- Cisco IOS translation rules
- Cisco CME dial plan pattern commands

Another way to configure the mapping from PSTN numbers to internal extensions is by using the E.164 number configuration of Cisco CME and Cisco UE. These fields essentially associate a second numberthe fully qualified E.164 numberwith the extension or mailbox so that dialing either number terminates on the correct extension or mailbox.

Cisco CME has a Secondary Number field, as shown in Example 10-10. You can access this same field by using the GUI by navigating to the Configure > Extensions screen. Filling in the Secondary Number field on that screen associates an additional number with the extension so that calls dialing the extension directly (likely other IP phones) and calls dialing the DID number (likely PSTN callers) terminate on the phone without further dialing plan assistance.

Example 10-10. Cisco CME Secondary Number Field

```
Router#show running-config
ephone-dn  1
 number 6001 secondary 5103953001
 description Grace Garrett
 name Grace Garrett
 call-forward busy 6800
 call-forward noan 6800 timeout 10
```

The effect of configuring the Secondary Number field is that an E.164 PSTN number can terminate directly on an extension without digit manipulation of the called digits. If this call forwards into Cisco UE voice mail, the Called Digits field still contains the full E.164 number, which must be recognized by Cisco UE as a mailbox owner. This requires the configuration of the Cisco UE E.164 field.

Cisco UE has a Primary E.164 Number field that you can also access by navigating to the Configure > Users screen in the GUI and clicking a specific user to highlight that person's parameters. Filling in this field associates an additional number with the user (and therefore the user's mailbox) so that calls dialing this number directly can enter the mailbox for the correct user.

Voice Mail Networking

Cisco UE release 2.0 introduced basic voice mail networking between systems using blind addressing, and release 2.1 enhances this functionality with spoken-name and limited dial-by-name addressing using a local directory and cache. You can network voice mail messaging between Cisco UE systems at different sites, as well as with Cisco Unity systems, using Voice Profile for Internet Mail (VPIM) connectivity between the systems. Cisco UE does not support networking with any voice messaging systems other than Cisco UE and Cisco Unity systems.

You can configure Cisco UE networking using either Domain Name System (DNS) host names or explicit IP addressing. If you are networking with a Cisco Unity system, you must use a DNS configuration.

The following sections provide a brief overview of Cisco UE voice mail networking operation, including the following topics:

- Applicable standards
- Definition of network locations
- Voice mail network location addressing
- Directories
- Network broadcast messaging
- Message formats
- Nondelivery notification

Standards

Voice mail networking uses the following Internet Engineering Task Force (IETF) standards:

- RFC 3801, Voice Profile for Internet Mail - Version 2 (VPIMv2)
- RFC 2821, Simple Mail Transfer Protocol (SMTP)
- RFCs 20452049, Multipurpose Internet Mail Extensions (MIME) Part One: Format of Internet Message Bodies
- RFC 822, Standard for the Format of ARPA Internet Text Messages

The Simple Mail Transfer Protocol (SMTP) and Multipurpose Internet Mail Extension (MIME) protocols are the basic protocols used to send and receive e-mail. Voice mail networking leverages the exact same protocols and infrastructure. The primary attributes of a voice mail (compared to an e-mail) are the address (voice mail uses a phone number and e-mail uses a user ID) and the message's payload type (voice mail uses an encoded form of a .wav file

Voice Mail Deployment Considerations

Cisco IPC Express deployment options were covered in Chapter 2, "Building a Cisco IPC Express Network." Three primary deployment models with Cisco CME and voice mail using Cisco UE were discussed:

- Standalone office
- Multisite business or enterprise
- Service provider managed multisite network

You can deploy Cisco CME without Cisco UE, either completely without an AA or voice mail, or with alternate voice mail solutions, as covered in Chapter 11. However, Cisco UE was specifically designed to provide these applications with Cisco CME. It offers the greatest breadth of features and the highest level of integration.

You can choose to deploy Cisco UE at only some sites in your network, while the remaining sites have either no voice mail or have voice mail provided by Cisco Unity or an alternate system. As explained in the preceding section, Cisco UE can network voice messages with other Cisco UE sites as well as with Cisco Unity sites.

Before Cisco UE release 2.0, if Cisco UE and Cisco CME were deployed together at a site, they had to be present in the same router chassis. As of Cisco UE release 2.0, you can house Cisco UE on a router that is physically separate from the one that provides Cisco CME call processing services to your site. This is not a likely or preferred deployment option, but in some cases, you may have an existing router you want to redeploy, and it does not have enough slots to add Cisco UE into the chassis. In this case, you can run Cisco CME on one router and physically house Cisco UE on a different router at the same site.

Ensure that LAN connectivity exists between the two routers, however, as opposed to low-speed serial connectivity. Also, Cisco UE can provide voice mail to only a single Cisco CME site. You cannot "centralize" one Cisco UE system to provide voice mail to multiple remote Cisco CME sites, as you can do with a Cisco Unity system.

Summary

This chapter discussed the voice mail capabilities provided by the Cisco UE aspect of the Cisco IPC Express solution. The operation of the voice mail features was discussed from a subscriber, caller, and administrator point of view. System-level features such as storage options, as well as typical voice mail activities, such as sending and addressing messages, were also covered.

This chapter explained how to configure Name fields in Cisco CME and Cisco UE to tailor the phone displays, dial-by-name options, and directories in your office. The configuration and operation of voice mail networking between sites was discussed to help you network your sites into a cohesive business.

You should now have a good understanding of the features Cisco UE provides and how to tailor the system to your needs for call flows, name displays, and the general operation of your office. The internals of the application and how to troubleshoot potential problems are covered later in this book.

Chapter 11. Cisco CME External Voice Mail Options

This chapter covers the following topics:

- Cisco Unity voice mail
- Stonevoice voice mail
- Analog voice mail
- Public Switched Telephone Network (PSTN)-based voice mail

Cisco Unity Express (UE) voice mail was covered in Chapter 10, "Cisco IPC Express Integrated Voice Mail." It represents the integrated voice mail system recommended for use with Cisco CallManager Express (CME). However, you also have the option to deploy one of several external voice mail systems with Cisco CME. You might want to consider one of these voice mail options for your office in the following situations:

- If you are using a Cisco CME platform, such as the Cisco 1760-V, that does not support Cisco UE (although you could use a separate router to house Cisco UE as of release 2.0)
- If your Cisco CME router platform does not have any available slots to add the Cisco UE hardware
- If you require features such as unified messaging that are not yet available with Cisco UE
- If you want to deploy a centralized voice mail system to support multiple Cisco CME sites instead of a distributed voice mail option at each site
- If you have an existing legacy voice mail system that you want to continue to use

This chapter briefly reviews the external voice mail options available for use with Cisco CME. These options include the Cisco Unity voice mail system and several non-Cisco systems that integrate with Cisco CME via either H.323 voice over IP (VoIP) or an analog phone interface.

Cisco Unity Voice Mail

Cisco Unity is a Windows 2000 server-based IP unified messaging system. Cisco Unity scales up to several thousand users and typically is deployed in a central or campus site of an enterprise network with a Cisco CallManager providing the call control.

Note

Note that Cisco Unity and Cisco UE are two different voice mail systems. Cisco UE is a hardware module installed inside the Cisco CME router scaling up to 100 voice mailboxes, whereas Cisco Unity is a separate Windows server platform scaling up to thousands of users.

Cisco Unity's unified messaging capabilities allow you to integrate voice mail, e-mail, and faxes into the same end-user mailbox. The mailbox operation is highly customizable via call handlers. Cisco Unity provides options for integrating with the Microsoft Outlook or Lotus Notes mail architectures. Cisco Unity leverages Active Directory to access your network's user and location directory.

Although Cisco Unity's unified messaging features provide many productivity-enhancing applications such as Cisco Personal Assistant and text-to-speech support, you can also choose to deploy it as a voice mail-only system. Cisco Personal Assistant is a telephony application suite that streamlines communications by helping users manage how and where they can be reached.

Different levels of licensing are available with Cisco Unity for a voice mail-only deployment or a full unified messaging system. Cisco Unity is a sophisticated messaging system with robust failover and networking options. A full discussion of its capabilities is beyond the scope of this book, but you can consult Cisco.com and search for "Cisco Unity" to find more information.

Cisco Unity uses IP as the transport and Skinny Client Control Protocol (SCCP) for call control, appearing to Cisco CME as an IP phone (SCCP) endpoint. Cisco Unity can support a single Cisco CME in a standalone deployment or can be deployed at a central site in your network with multiple Cisco CMEs at remote sites in a centralized voice mail scenario. These architectures are discussed further in the following sections.

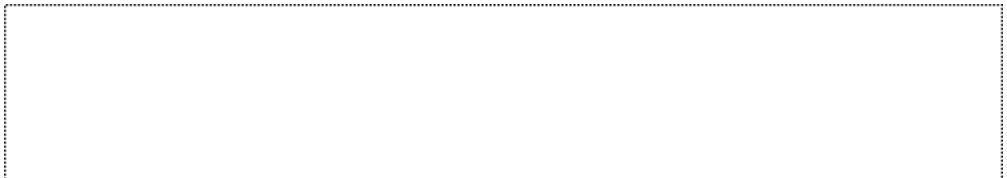
Note

Cisco CME is Cisco IOS-based, whereas Cisco Unity is a Windows 2000-based application; therefore, users considering this combination of applications should have a good working knowledge of both operating systems. Cisco Unity uses Microsoft Exchange (5.5, 2000, and 2003) or Lotus Domino mail stores, so familiarity with these technologies is also beneficial.

Standalone Cisco CME System with Cisco Unity

You can connect a standalone Cisco CME with a dedicated Cisco Unity system to provide unified or voice mail services to a single site. Though technically feasible, this is often not a cost-effective way to deploy Cisco Unity. Cisco Unity is an application designed to support large numbers of users, whereas Cisco CME can support only up to 240 users. Figure 11-1 shows how Cisco CME is connected to the Cisco Unity system.

Figure 11-1. Standalone Cisco CME with Cisco Unity Messaging

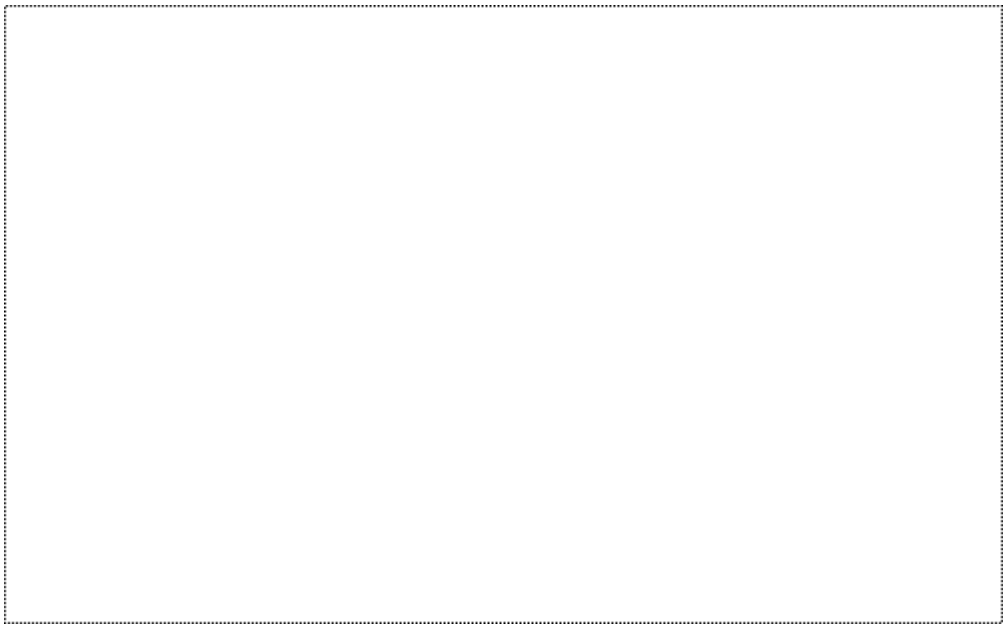


Stonevoice Voice Mail

Cisco CME can integrate with various non-Cisco voice mail systems using H.323. One of the H.323 voice systems supported by Cisco CME is the Stonevoice Switch Answering Machine (SSAM), a unified messaging system designed to provide access to and control over software-based voice mail services.

SSAM is a Windows 2000-based application that runs on an external PC. All traffic between Cisco CME and SSAM uses H.323. Figure 11-4 shows how the Stonevoice SSAM application integrates with Cisco CME.

Figure 11-4. Cisco CME with Stonevoice SSAM Voice Mail



When integrated with CME, SSAM supports the following:

- Direct access to voice mail
- Call forward no answer (CFNA) or call forward busy (CFB) to a personal greeting
- MWI

For more information on the SSAM system, go to http://www.stonevoice.com/.

The following sections provide more details on integrating a Stonevoice system with Cisco CME, including

- Configuring Cisco CME for Stonevoice
- MWI from Stonevoice

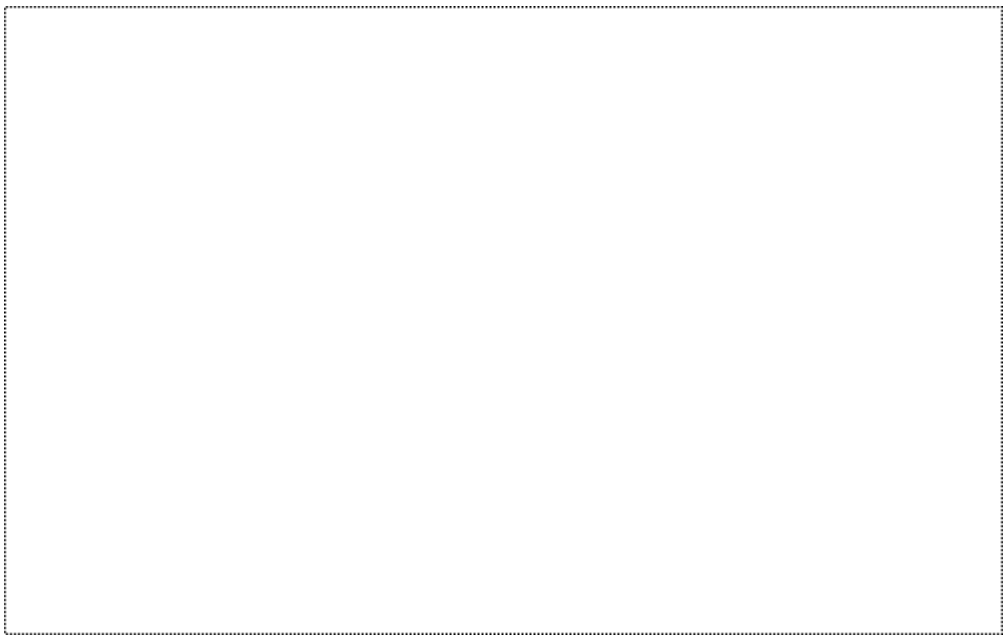
Configuring Cisco CME for Stonevoice

Communication between Cisco CME and SSAM is via H.323, so you have to configure an H.323 dial peer to direct calls into the SSAM system. You must configure a voice mail pilot number (for example, 9999) on SSAM for

Analog Voice Mail

You can integrate Cisco CME with analog systems to provide voice mail services, as shown in Figure 11-5. In general, these systems connect to the Cisco CME via Foreign Exchange Station (FXS) analog phone interfaces. Each port is configured as a normal plain old telephone service (POTS) dial peer in Cisco CME.

Figure 11-5. Cisco CME with Analog Voice Mail



Cisco CME interacts with the analog voice mail system via inband dual-tone multifrequency (DTMF) tones. All call routing and MWI information exchanged between Cisco CME and the voice mail system also occurs via DTMF tones.

When integrated with Cisco CME, an analog voice mail system provides the following:

- Direct access to voice mail
- CFNA or CFB to a personal greeting
- MWI

Many types of analog voice mail systems are available. The Octel system from Avaya and the Reception system from Active Voice, LLC are two of the more popular models. The following sections discuss Cisco CME integration with these systems.

Octel

Integrating the Octel voice mail system with Cisco CME requires configuration on both systems. The configuration sample in Example 11-8 shows how to configure the CME.

Example 11-8. Cisco CME Configuration for Octel Analog Integration

```
router#show running-config
call application voice bator flash:app-h450-transfer.2.0.0.9.tcl
```

PSTN-Based Voice Mail

Another option for voice mail with Cisco CME is through your PSTN provider. The call flows to the voice mail from your PSTN provider are controlled by the central office (CO) PSTN switch. If the lines to your business are busy or don't answer, the voice mail system at the CO picks up and stores the senders' voice messages at the voice mail storage located at the CO. You do not need to configure Cisco CME for this type of voice mail. You do need to configure MWI, however.

CO-based voice mail systems signal MWI by using stutter dial tone. When you go off-hook on your phone, you hear the tone and know you have a message. This works well if you use an analog phone directly connected to the CO and you, therefore, get dial tone from the CO. If you have a Cisco CME system in your office with IP phones on, however, dial tone comes from the Cisco CME system, not from the CO.

To hear stutter dial tone provided by a CO-based voice mail system, you can use the Cisco CME 3.2 FXO Trunk Line Select feature by pressing a button on your IP phone. It directly selects a CO Foreign Exchange Office (FXO) line, which gets dial tone from the CO, and you can hear stutter dial tone.

Example 11-10 shows how to use the trunk command to create a direct connection to a CO line on an IP phone.

Example 11-10. FXO Trunk Command

```
router#show running-config
voice-port 1/0/0
  connection plaropx 1082
dial-peer voice 82 pots
  destination-pattern 82
    port 1/0/0
    forward-digits 0
ephone-dn 10
    number 1010
    name manager
ephone-dn 11
  number 1082
    name private-line
  trunk 82
ephone 1
    button 1:10 2:11
```

Example 11-10 shows the following sequence of events:

- You press the softkey 2 on your phone, and it automatically triggers the configuration for ephone-dn 11.
- ephone-dn 11 initiates a call to 82, which matches the dial peer that points to FXO line 1/0/0.
- When line 1/0/0 goes off-hook (as if you had picked up an analog phone set directly connected to the CO), it does not dial any digits. The effect is that you can hear the dial tone provided by the CO.

Summary

In this chapter, you learned about several external voice mail options you can use with your Cisco CME system if you choose not to use the integrated voice mail application provided by Cisco UE.

The Cisco Unity unified messaging system provides a very good voice mail alterative, especially if you have a multisite network that has a larger location with several smaller locations, and you want to centralize voice mail for all locations at one site.

Non-Cisco voice mail solutions, such as Stonevoice, provide voice mail applications that integrate via H.323 with Cisco CME. Another alternative to integrate with some popular voice mail systems, such as Octel and Active Voice Reception, is to use analog connectivity and DTMF interaction between Cisco CME and the voice mail system. Using PSTN-based voice mail is also an option.

This chapter covered the trade-offs of the different methods and provided brief configuration information on how to set up your Cisco CME system to interwork with external voice mail systems.

Chapter 12. Additional External Applications with Cisco CME

This chapter covers the following topics:

- Telephony Application Programming Interface (TAPI) and XML application architecture
- TAPI applications
- XML applications

In addition to the voice mail and automated attendant (AA) applications discussed in Chapters 9, 10, and 11, Cisco CallManager Express (CME) can provide even more capabilities by integrating with external applications offered by Cisco and other vendors. This chapter briefly reviews the external applications that you can integrate with Cisco CME, such as applications that use the Telephony Application Programming Interface (TAPI) and Extensible Markup Language (XML) interfaces.

Some external applications integrate with Cisco CME by using Skinny Client Control Protocol (SCCP) connections. This chapter does not discuss these application because they appear to Cisco CME as IP phones (SCCP endpoints) and not as an application using an application interface, such as TAPI or XML. SCCP applications register with Cisco CME as IP phones do and communicate with the Cisco CME call processing features via SCCP messages as any IP phone would. Applications in this category include softphone PC applications such as the Cisco IP Communicator and IP Blue Software Solutions' softphone. More information about IP Blue can be found at http://www.ipblue.com/.

TAPI and XML Application Architecture

TAPI applications interface directly with the Cisco CME call processing software to control the call processing signaling events that apply to an IP phone. For example, a TAPI application can answer a call on behalf of a phone, make the phone go off-hook, or disconnect a call. Windows-based screen-pop applications typically use TAPI.

XML applications interface directly with the IP phone and leverage its HTTP capabilities. The phone has one or more URLs as part of its software load that it accesses when buttons such as the services or directory keys are pressed. The XML application renders some text or graphics on the phone's display. Cisco CME call processing is not involved in this application interchange. XML applications control the phone display while the phone is idle (that is, not on a call).

Figure 12-1 shows how TAPI and XML applications interface with Cisco CME and its IP phones. The left side of the figure depicts a TAPI application. An employee has an IP phone and PC on the desktop. The PC runs a productivity application, such as a contact management application, that provides a screen pop based on caller ID whenever a call starts ringing. All messaging between the application and the phone passes through Cisco CME and is interpreted by the Cisco CME software. On the right side of the figure is an XML application. The phone has a URL that points to a server. The server application writes to the phone display using HTTP.

Figure 12-1. TAPI and XML Application Architecture

[View full size image]

TAPI Applications

TAPI is a Microsoft software application interface for integrating-telephony-services into Windows-based PC applications. Cisco CME provides telephony services via a TAPI Service Provider (TSP) interface to applications.

The TSP allows TAPI-based applications such as Microsoft Outlook and Microsoft Customer Relationship Management (CRM) to provide call control to the IP phones connected to Cisco CME. Other TAPI-based applications are available in the industry, such as automatic dialers. You can use these applications to control an IP phone to make and receive calls via a computer or to trigger database lookups based on caller ID.

The following sections describe TAPI in more detail, including the following topics:

- The Cisco CME TAPI Light implementation and capabilities
- Cisco CME TSP functions
- Cisco CRM Communications Connector

Cisco CME TAPI Light

Cisco CME offers a TAPI Light capability, which is not a full TAPI implementation but a selection of the applicable components for Cisco CME. The implementation consists of two parts: one part resides on the Windows platform, and the other part resides in Cisco CME's Cisco IOS software.

The interface between the TSP in the Windows application and Cisco CME uses SCCP over TCP. Cisco CME listens on a standard TCP port, while the TAPI client authenticates to Cisco CME by providing a username and password unique for each IP phone on Cisco CME. The Windows application's TSP must have the same username, password, and port number configured to be able to connect successfully with Cisco CME and exert phone and call control. The username and password authentication provides a layer of security to Cisco CME to enable authorized application development.

Example 12-1 shows the configuration of the username and password associated with the IP phone on Cisco CME. This information must be quoted by the TAPI application during login to be able to control the phone. The telephony-service ip source-address command specifies the port number used for communication between Cisco CME and the TAPI application.

Example 12-1. IP Phone Configuration

```
router#show running-config
telephony-service
 ip source-address 172.19.153.129 port 2000
!
ephone-dn  1
 number 3001
 description User1
 name User1
 call-forward busy 3105
 call-forward noan 3105 timeout 10
!
ephone  1
 username "User 1" password user1
 mac-address 0009.B7F7.5793
 speed-dial 4 3100 label "AA"
 button  1:1
```

Extensive Markup Language Applications

XML is a text markup language designed to control web-based documents. With XML, you can create web pages customized for specific application requirements.

This section briefly discusses the XML applications applicable to Cisco CME IP phones. For more information on developing XML applications, go to http://www.cisco.com/en/US/products/svcs/ps3034/ps5408/ps5418/serv_home.html, or go to Cisco.com and search for "Developer Support Central."

General XML Phone Services

XML services on Cisco IP phones give you another way to perform or access more business applications. Some examples of XML-based services on IP phones are user direct-dial directory, announcements, and advertisements.

The IP phones are equipped with a pixel-based display that can display full graphics instead of just text on the screen. The pixel-based display capabilities allow you to use sophisticated graphical presentations for applications on Cisco IP phones and make them available at any desktop, counter, or location. In addition, you can select prepackaged applications. A sample phone display with applications is shown in Figure 12-2.

Figure 12-2. XML Application on a Cisco IP Phone Display

Cisco CME XML Phone Services

In a Cisco CME application, XML between an XML server and the IP phones provides customized phone displays and services. The interaction between the IP phone and the XML server includes the following events:

1. The IP phone sends an HTTP request to the application server.
2. The server renders an XML document and sends it to the IP phone.
3. The IP phone parses the XML document and renders the screen graphics on the IP phone display.

You can build applications particular to your business, such as a store inventory or stock quote lookup capability. These applications are especially useful to employees who do not have PCs or do not work at desks. You can also write more general applications, such as displaying a large-numbered clock on the IP phone (that displays when the phone is idle), and use this in conference rooms, lobbies, or break rooms instead of a wall clock.

You can find more information about XML-based IP phone services and productivity applications at the Cisco Applications Central web portal. To get to this site, go to Cisco.com and search for "IP Communications Applications Central." This site also has application partner information and discussion forums.

XML Application Example

Summary

This chapter provided a brief overview of the external TAPI and XML applications you can use with Cisco CME. With these applications you can tailor the native capabilities of Cisco CME IP phones much more closely to your own business environment. This chapter discussed the architecture necessary to activate the applications, the required Cisco CME configuration, and some examples of applications.

Part III: Administration and Management

Chapter 13: Cisco IPC Express General Administration and Initial System Setup

Chapter 14: Configuring and Managing Cisco IPC Express Systems

Chapter 15: Cisco IPC Express System Configuration Example

Chapter 13. Cisco IPC Express General Administration and Initial System Setup

This chapter covers the following topics:

- An overview of administrative access to a Cisco IP Communications (IPC) Express system
- System installation and initial setup
- Cisco Call Manager Express (CME) graphical user interface (GUI) customization via XML
- Cisco Zero Touch deployment

The chapters in Part II, "Feature Operation and Applications,"discussed Cisco IP Communications (IPC) Express system capabilities and features to implement a complete IP communications system for your office. The chapters in Part III, "Administration and Management," look at the configuration and administration of a Cisco IPC Express system.

This chapter focuses on the administration methods and access to the system, as well as the installation and initial setup of a new system. Chapter 14, "Configuring and Managing Cisco IPC Express Systems," offers a more in-depth view of configuring the system's features. Chapter 15, "Cisco IPC Express System Configuration Example," takes you through a step-by-step configuration setup of a sample system.

Cisco IPC Express is composed of four major components, as described in Chapter 1, "Introducing Cisco IPC Express":

- The router platform, which provides IP connectivity to the network and connectivity to the Public Switched Telephone Network (PSTN)
- Cisco Call Manager Express (CME), which provides call processing capabilities
- Cisco Unity Express (UE), which provides automated attendant (AA) and voice mail capabilities
- IP phone endpoints

In this chapter, the IP endpoints are considered part of Cisco CME and, therefore, are covered in the Cisco CME sections along with the call processing features provided on the phones.

Note

Cisco Unity Express (UE) is an optional part of your system. If you do not have this component installed, you can skip the sections in this chapter particular to Cisco UE.

A comprehensive, detailed discussion of all Cisco IPC Express system administration capabilities would take a book in itself. Therefore, this chapter introduces the salient aspects of Cisco IPC Express system administration interfaces,

Administrative Access Overview

Cisco IPC Express is a converged office communications system deployed by several different types of businesses:

- Small and medium businesses with one or more sites (using the standalone or multisite deployment scenarios, as discussed in Chapter 2, "Building a Cisco IPC Express Network")
- A branch office of an enterprise network of any size (also using the standalone or multisite deployment scenarios)
- A customer premises equipment (CPE) solution offered by service providers (SPs) as a managed service

These different ways in which you can deploy Cisco IPC Express require different types of administrative interfaces and different levels of access. For that reason, a Cisco IPC Express system offers a full command-line interface (CLI), a browser-based graphical user interface (GUI) for the key features, and several setup wizards to expedite system initialization tasks.

System installation and setup always require CLI access to install and provision enough of the system to be able to drive a GUI. You can set up the general call processing, AA, and voice mail features using either the CLI or the GUI. End-user interaction with voice mailboxes uses a Telephony User Interface (TUI), which means that users use a phone to interact with the system.

User IDs, logins, and passwords are defined at various levels. Different types of administrator accounts exist to partition access to different aspects of the system.

Cisco CME also offers several application programming interfaces (APIs) to interface with external applications for monitoring, configuration, and end-user phone applications.

Command-Line Interface

CLI is typically preferred by large enterprises and managed services networks where hundreds or thousands of systems must be provisioned. These are often scripted by higher-level network management systems centralized in a data center or a network operations center (NOC). Resellers and system integrators often use the CLI for initial system setup before they bring the system to your office. For the expert user, CLI access is often more expeditious than using a GUI.

The CLI for Cisco CME is part of the Cisco IOS router CLI. Access to the Cisco UE CLI requires that you log into the router CLI and then open a session to the Cisco UE application's CLI, which is modeled on Cisco IOS CLI but is not exactly the same. If you're a system administrator and are familiar with Cisco IOS and CLI commands, you can Telnet to the Cisco CME router (or use the console port), access the Cisco UE service engine from the same Cisco CME router, and perform all the setup and configuration tasks by using CLI commands.

Users unfamiliar with CLI in general and Cisco IOS router CLI in particular might find it much more intuitive to perform day-to-day moves, adds, and changes using the system GUI after an administrator or system reseller has set up the initial system for you and created a GUI account.

The following sections briefly summarize the Cisco CME and Cisco UE CLI commands.

Cisco CME CLI Command Summary

General Cisco CME CLI commands are under the telephony-service command on the router. Other key Cisco CME commands are ephone and ephone-dn, where many of the phone and call processing features are configured. You can find a more in-depth description of all the Cisco CME commands in the Cisco CME Administrator and

System Installation and Initial Setup

When you order a new Cisco IPC Express system, both the router and the Cisco UE hardware come preinstalled with the software and licensing you purchased. You do not need to do a software installation for a new system unless you have to upgrade the software levels from what was originally ordered. This section provides a high-level overview of the software installation and initial system setup process.

Although you do not need to install software on a new system, you must configure the system to make it functional. Several setup utilities can help you with these first-time tasks and these utilities, as discussed in the following sections.

Software Installation

Cisco CME software is part of the general Cisco IOS router software. You need an IP voice or greater Cisco IOS image to get voice features, including Cisco CME. In addition to the Cisco IOS software, you purchase the following licenses based on the number of users at your site, as covered in Appendix A, "Cisco IPC Express Features, Releases, and Ordering Information":

- Cisco CME seat licenses for the number of IP phones
- Cisco UE voice mail licenses for the number of mailboxes

Note

If you are upgrading an existing router to become a Cisco CME system, you have to follow a full software installation procedure. Also, if you are upgrading an existing Cisco UE module to a new level of licensing or software, a software installation is required. Refer to Cisco.com for more detailed step-by-step instructions on router, Cisco CME, and Cisco UE software installation procedures.

Cisco CME Router Software Installation

Installing Cisco CME is exactly the same as installing your router with software. In summary, here are the steps:

Step 1. Download the correct Cisco IOS software image and release level from the Cisco.com Software Center at http://www.cisco.com/go/software. Note that you have to be a registered Cisco.com user to access the Software Center.

Step 2. Place these files on a Trivial File Transfer Protocol. (TFTP) or FTP server at your site.

Step 3. Use the Cisco IOS router CLI to load the software image into router Flash.

Step 4. Reboot the router.

In addition to loading the Cisco IOS software image, several processes apply to Cisco CME installation:

- Download the phone images into the router Flash.
- Download the GUI files into the router Flash. (You can use the archive command on the router to extract the contents of the entire .tar file with a single step into the router Flash.)
-

Cisco CME GUI Customization Via XML

You can customize Cisco CME GUI features to provide a customer administrator login. This Cisco CME facility is used if you as a reseller are setting up a system for your end customer and that person wants to perform certain administrative functions on the system while you remain responsible for other items. You can use the system administrator login and access all system GUI features. You can limit the items your end customer sees in the GUI (using the customer administrator login) to those he or she is responsible for managing.

The following sections discuss how to customize the Cisco CME GUI features, and change the look and feel of the GUI pages with a sample XML file. The Cisco UE GUI cannot be customized in this manner.

The Cisco CME system administrator defines what a customer administrator can do by defining and setting the tags in an XML file. The XML file can be read in through the CLI to control the set of GUI functions allowed for a customer administrator.

Based on the existing Cisco CME XML Schema Template file, you might create a sample.xml and copy it to your router's Flash. In the sample.xml file, you can use tags with values of Show or Hide to allow or deny the customer administrator access to the corresponding GUI features.

If a tag item is missing from the XML file, the default behavior is to show. The customer administrator's access is equivalent to the system administrator under the following circumstances:

- No XML file exists in Flash
- After a no web customize load command is executed on the router (this command can be written to nonvolatile RAM [NVRAM])

You configure Cisco CME to allow GUI customization using the sample.xml file as shown in Example 13-20.

Example 13-20. Using an XML File for GUI Customization

```
telephony-service
   web admin system name admin password admin
   web admin customer name Cisco password Cisco
  web customize load sample.xml
```

Cisco CME XML Schema Template

The Cisco CME XML Schema Template is shown in Example 13-21. This sample template lists all the GUI features available to be customized using XML tags. You can select each to hide or show. By default, all features are shown in the GUI pages if neither Hide nor Show is specified.

Example 13-21. Cisco CME XML Schema Template

```
<Presentation>
  <MainMenu>
     <!-- Take Higher Precedence over CLI "dn-wed-edit" -->
     <AddExtension> [Hide | Show] </AddExtension>
     <DeleteExtension> [Hide | Show] </DeleteExtension>
     <AddPhone> [Hide | Show] </AddPhone>
     <DeletePhone> [Hide | Show] </DeletePhone>
  </MainMenu>
  <Extension>
      <!-- Control both view and change, and possible add or delete -->
      <SequenceNumber> [Hide | Show] </SequenceNumber>
```

Cisco Zero Touch Deployment

Cisco Networking Services (CNS) technology provides the infrastructure for automated configuration of large numbers of network devices. Based on CNS event and configuration agents, it eliminates the need for an on-site technician to initialize the devices. The CNS Zero Touch feature provides a deployment solution in which the router contacts a CNS CE to retrieve its full configuration automatically. This capability is made possible through a single generic bootstrap configuration file common across all service provider end customers subscribing to the services. Within the CNS framework, customers can create this generic bootstrap configuration without device-specific or network-specific information, such as interface type or line type.

Understanding Cisco Zero Touch Deployment Components

Cisco Zero Touch deployment consists of the following three components:

- Cisco Configuration Express (CX)
- Cisco CNS Configuration Engine (CNS CE)
- Cisco CNS Agent built into Cisco IOS software

Cisco Configuration Express

Cisco Configuration Express (CX) is an online ordering system and customizable inline manufacturing process that lets SPs easily deploy customer premises equipment (CPE)-based managed services to their small-to-medium sized business and enterprise customers. When ordering Cisco products, SPs use CX to specify the shipping instructions, including the Cisco IOS software version and a bootstrap configuration, which are configured, tested, and shipped with the CPE. The resulting fully configured CPE is shipped either directly to the end customer site or to the service provider's warehouse.

The bootstrap configuration integrates with Cisco CNS CE the moment the CPE devices are plugged into the network at the end-customer site.

Cisco CNS Configuration Engine

Cisco CNS CE runs on the Cisco CNS 2100 series Intelligence Engine (CNS IE 2100) hardware platform as well as customer UNIX servers. It is a secure and scalable deployment and configuration management application that provides an intelligent network interface to applications and users supporting up to 5000 Cisco CPE devices.

Cisco CNS CE includes the Configuration Service and Configuration Server. The Configuration Server communicates with the CNS Configuration Agent running on the managed Cisco CME via HTTP and transfers data in XML format parsed by the CNS Configuration Agent on the Cisco CME router using its own parser.

The CNS Configuration Service delivers device and service configurations to Cisco IOS devices for initial configuration and mass reconfiguration by logical groups. Routers receive their initial configuration from the CNS Configuration Service when they start up on the network the first time. The CNS Configuration Service uses the CNS Event Service to send and receive events required to apply configuration changes and to send success and failure notifications.

The templates created on the CNS CE are automatically pushed to the CPE devices running the bootstrap configuration.

For more information on Cisco CNS CE, go to Cisco.com, and search for "Cisco Configuration Engine."

Summary

This chapter reviewed the administrative interfaces to a Cisco IPC Express System, including CLI and GUI capabilities. Several types of logins control administrator and end-user privileges and access to the system. You can also customize the Cisco CME via XML files and style sheets to fit your business model.

Cisco CME and Cisco UE system installation and various setup tools and utilities were discussed. For SPs or large enterprises managing hundreds or thousands of systems, the Cisco Zero Touch deployment mechanism with Cisco IE 2100 was covered.

Chapter 14. Configuring and Managing Cisco IPC Express Systems

This chapter covers the following topics:

- Cisco IP Communications (IPC) Express system graphical user interface (GUI) overview
- Configuring the router
- Configuring IP phones and extensions
- Configuring Public Switched Telephone Network (PSTN) interfaces
- Configuring extensions and the dial plan
- Configuring Cisco CallManager Express (CME) call processing features
- Configuring the Cisco Unity Express (UE) automated attendant (AA)
- Configuring Cisco UE voice mail
- Configuring the Cisco UE Administration via Telephony (AVT)
- Configuring Cisco UE Backup and Restore
- Configuring Cisco CME interconnection with other sites
- Security best practices for Cisco CME
- Security best practices for Cisco UE
- Configuring and monitoring via network management systems using XML Layer (AXL)/Simple Object Access Protocol (SOAP)
- Managing Cisco IPC Express systems by managed services and enterprises
- Cisco voice network management solutions
-

Cisco IPC Express System GUI Overview

The overall aspects of a Cisco IPC Express system that you must configure include the following, many of which are optional:

- Router
- IP phones and lines
- Public Switched Telephone Network (PSTN) interfaces
- Extensions and the dial plan
- Cisco CallManager Expess (CME) call processing features such as intercom and hunting
- automated attendant (AA)
- Personal mailboxes and general delivery mailboxes (GDMs)
- Message waiting indicator (MWI)
- Voice mail networking
- Backup and restore

The Cisco IPC Express GUI does not provide access to all the system's components. You must configure many basic router features (such as IP routing, Dynamic Host Configuration Protocol [DHCP], and virtual LAN [VLAN]), the PSTN trunk interfaces, many dial plan aspects, intersite interconnection, and security features on the router itself. The router leverages the Cisco IOS general and voice infrastructure software discussed in Chapter 3, "Cisco IPC Express Architecture Overview."

You can configure components such as IP phones, extensions, AA, and voice mailboxes via the Cisco IPC Express GUI. This section briefly introduces the GUI. Later sections provide a configuration overview of each system component.

Figure 14-1 shows the following main menu items of the Cisco IPC Express GUI:

- Configure
- Voice Mail
- Administration

Configuring the Router

Cisco IPC Express is based on the router architecture. Many aspects of the system require certain router and LAN switch components to be configured and operational:

- IP interfaces such as Ethernet, serial WAN, or internetworking interfaces
- IP addressing and routing
- DHCP
- Router login and password accounts and security measures (such as authentication, authorization, and accounting [AAA]/Remote Authentication Dial-In User Service [RADIUS])
- NTP for clock coordination
- Network Trivial File Transfer Protocol (TFTP) access to install and reload the router
- Voice and data virtual LANs (VLANs) and inline power (if supported and used)
- Security features such as network address translation (Network Address Translation [NAT]) and firewall

In addition to these items, you must configure the basic aspects of Cisco IPC Express:

- Turn on Cisco CME with the telephony-service command.
- If Cisco UE is present, configure the service engine interface.
- Configure HTTP access for the GUI.
- Install the Cisco CME GUI files into the router Flash.

Configuring IP Phones and Extensions

The features and configuration discussed thus far can be configured only via the router CLI because they cannot be accessed through the GUI. Cisco CME IP phones and extensions (lines) are the first features available using the GUI. After you successfully log into the Cisco CME GUI (as covered in Chapter 13), you might add, change, and remove ephone-dns, ephones, and speed-dial lists and perform other configuration tasks.

Note

Not all the Cisco CME features can be configured via the GUI. Some features might require router configuration (CLI), and others might depend on the privilege level granted with your login.

The following sections provide some highlights of the GUI configuration:

- Adding an extension
- Adding a phone
- Assigning an extension to a phone
- Changing or deleting an extension
- Changing or deleting a phone

For configuration details on more GUI features, choose Help > Configuration to launch the online GUI help, or access the Cisco CME and Cisco UE Administration Guides on Cisco.com.

Adding an Extension

An extension in the GUI maps to an ephone-dn in the router CLI. It is best to add the extensions before configuring the phones, because the phone's buttons have extensions assigned to them. Therefore, you can't complete the phone configuration until the extensions have been defined.

To add an extension to your system, choose Configure > Extensions and then click Add, as shown in Figure 14-2.

Figure 14-2. Adding an Extension

[View full size image]

Configuring PSTN Interfaces

You cannot configure the PSTN trunks of your Cisco CME system in the GUI. This requires CLI configuration on the router. The types of trunks and interfaces you can add to the system to connect to the PSTN or fax machines in your office were covered in Chapter 6, "Cisco CME PSTN Connectivity Options." Numerous configuration examples were also given in Chapter 6.

Adding PSTN connectivity involves the following configuration:

- The voice interface that connects to your PSTN trunks, analog phones, or fax machines
- The dial plan that directs calls to these interfaces

You might review the examples in Chapter 6 for more details on configuring a Foreign Exchange Office (FXO), Foreign Exchange Station (FXS), T1 Channel Associated Signaling (CAS), and Primary Rate Interface (PRI) Public Switched Telephone Network (PSTN) trunks and the dial plans to direct calls to these ports and trunks.

Configuring Extensions and the Dial Plan

The dial plan of your Cisco CME system is composed of the following components:

- Plain old telephone service (POTS) dial peers
- Voice over IP (VoIP) dial peers
- Extensions
- Digit manipulation features

The following sections describe configuring these components.

POTS Dial Peers

POTS dial peers are router configuration structures that point calls to a particular voice interface based on whether the dialed number matches certain criteria (as specified in the dial peer's destination-pattern). The voice interface can be a PSTN trunk, an analog phone, a fax machine, or an IP phone. Example 14-2 shows several sample POTS dial peers, including some for PSTN trunks and one for an analog phone or fax machine.

Example 14-2. POTS Dial Peer Examples

```
router#show running-config
!11-digit long-distance PSTN dialing with an access code of 9
dial-peer voice 1 pots
 preference 1
 destination-pattern 91..........
 port 2/0:23
 forward-digits 11
!
!7-digit local PSTN dialing with an access code of 9
dial-peer voice 4 pots
 destination-pattern 9[2-9]......
 port 2/0:23
 forward-digits 7
!
!Analog phone or fax machine
dial-peer voice 2701 pots
 destination-pattern 2701
 port 2/0/0
```

VoIP Dial Peers

VoIP dial peers are router configuration structures that point calls to a particular IP interface based on the same criteria that are used for POTS dial peers (that is, matching the dialed number to the dial peer's destination-pattern). IP interfaces can be H.323 or Session Initiation Protocol (SIP). Example 14-3 shows an H.323 dial peer that directs calls to another site where all the extensions start with three.

Example 14-3. VoIP Dial Peer Examples

```
router#show running-config
dial-peer voice 3000 voip
 destination-pattern 3...
```

Configuring Cisco CME Call Processing Features

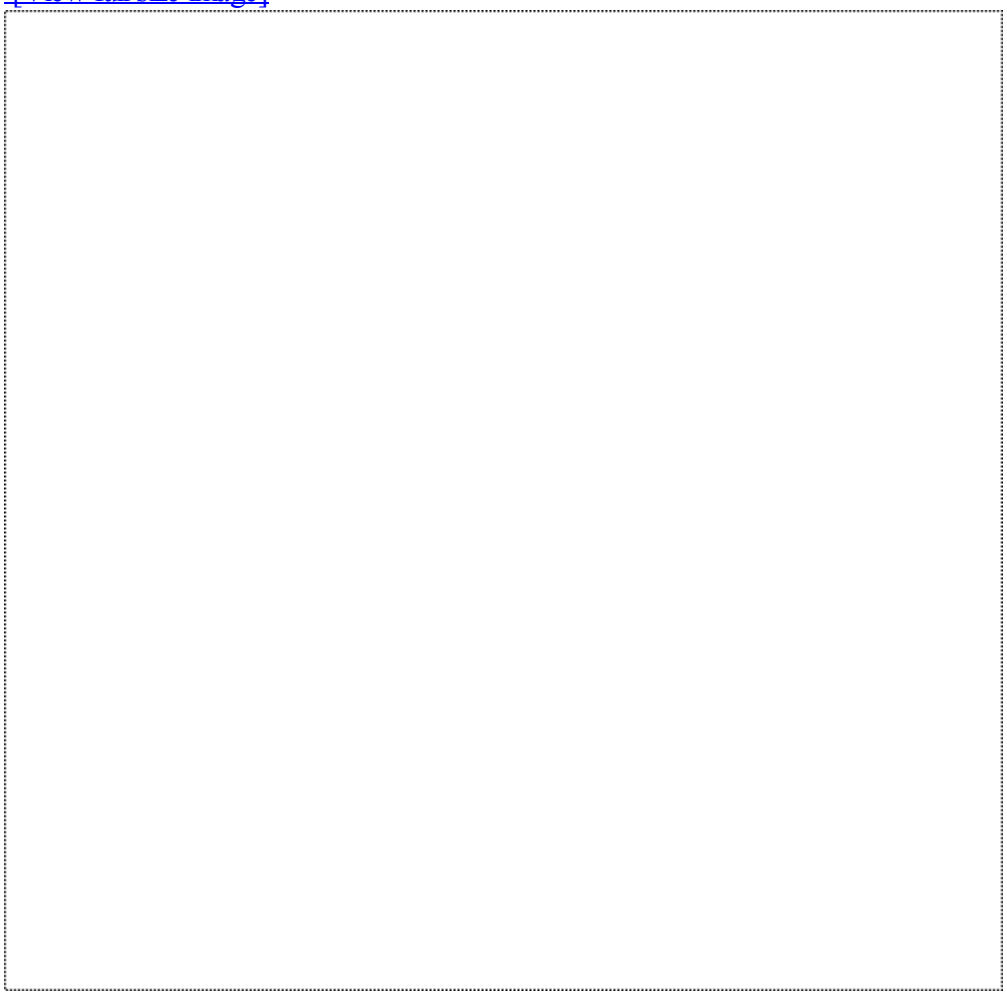
Chapter 5 discussed numerous Cisco CME call processing features. The following sections discuss how to configure the most common of these features. For further information, refer to the Cisco CME Administration and Configuration Guides on Cisco.com. You can configure all the features discussed in the following sections by using the GUI or CLI.

Configuring Speed Dial

On the Configure > Phones menu, scroll down to the speed-dial information. Here you might add or change the speed dials assigned to this phone, as shown in Figure 14-6.

Figure 14-6. Adding and Changing Speed-Dial Information

[View full size image]



Configuring Call Blocking Toll Bar Override with a PIN

From the Configure > System Parameters menu, select Call Blocking Configuration, as shown in Figure 14-7. You might add, change, or delete a block pattern (for example, 1900), and block date and day so that no calls to extension 1900 can be made during the date and time configured.

Figure 14-7. Configuring Call Blocking

[View full size image]

Configuring the Cisco UE AA

You set up the Cisco UE system-provided AA when you worked through the Initialization Wizard (see the section "Cisco UE Initialization Wizard" in Chapter 13). The configuration needed for the system AA includes the business hours, holiday list, operator extension, AA pilot number, and a few prompts spoken to callers by the system AA.

Configuring the System AA

Figure 14-19 shows the customization of the Cisco UE system AA. You start in the Voice Mail > Auto Attendant menu, and then click autoattendant in the list of AA scripts. This opens the AA Configuration Wizard. Go to the second step, Script Parameters, and set the parameter operExtn to the extension in your office where you want calls transferred when a caller chooses to speak to the operator. Figure 14-19 shows the operator extension set to extension 1001.

Figure 14-19. Cisco UE System AA Configuration

[View full size image]

To replace the system-provided prompts (shown in Figure 14-19), log into the AVT TUI (called the Greeting Management System [GMS] in Cisco UE releases before release 2.1), and record the following customized prompts for your office:

- welcomePrompt The welcome greeting spoken when the AA answers a call.
- busOpenPrompt The prompt following the welcome prompt, spoken if your office is currently open (based on the system time when the call arrives).
- busClosedPrompt The prompt following the welcome prompt, spoken if your office is currently closed (based on the system time when the call arrives).
- holidayPrompt The prompt following the welcome prompt, spoken if the current system date is listed as a

Configuring Cisco UE Voice Mail

Cisco UE offers both personal and GDMs, as covered in Chapter 10, "Cisco IPC Express Integrated Voice Mail." The following sections cover configuring the Cisco UE voice mail system, personal mailboxes, GDMs, and MWI. You also learn how to delete a mailbox.

Configuring the Voice Mail System

You have to set several system-level voice mail configuration parameters, including system attributes such as the voice mail pilot number, voice mail operator number, and mailbox system defaults. You also define individual mailboxes for users and groups.

Pilot Number

The voice mail pilot number is the extension subscribers call when they want to retrieve their voice mail. It is also the number that IP phones are forwarded to so that callers are deflected into voice mail to leave a message. The configuration of the pilot number is shown in Figure 14-25.

Figure 14-25. Voice Mail Pilot Number Configuration

[View full size image]

Voice Mail Operator

The voice mail operator is the extension number where callers are deflected if they leave a voice mail and do not hang up. This configuration is shown in Figure 14-25. The default voice mail operator is the AA.

System and Mailbox Defaults

When you create a new mailbox for an existing user, or when you add a user at the same time as the mailbox, you can use the default system mailbox parameters. As the administrator, you can override the values of these parameters for each mailbox at creation time or later, or you can reset the system defaults so that all newly created mailboxes are assigned the same parameters.

You should set the per-mailbox parameters carefully before creating a large number of user mailboxes. When the system defaults are changed, existing mailboxes are unaffected. The default settings apply only to creating new mailboxes. For example, if the message expiry time default parameter is set to 30 days, all mailboxes created are assigned a 30-day expiry time. If you create 25 mailboxes before deciding that the expiry time should instead be 60 days, you then have to change each of the 25 existing mailboxes individually to reset the expiry time to 60 days. Changing the default value to 60 days affects only new mailboxes you create after you change the default value. The per-mailbox parameters are set in the Defaults > Mailbox window, shown in Figure 14-26. They include the

Configuring the AVT

You can manage Cisco UE AA prompts, broadcast messaging, spoken names for remote users and location, and the Emergency Alternate Greeting (EAG) by using the TUI. This means that the administrator can dial in on a phone and make small changes to the system. The AVT requires an extension and a PIN to log in. It authenticates users based on the extension/mailbox PIN combination. It also checks that the user associated with the extension/PIN combination has administrative privileges on the system.

Configuring Cisco UE Backup and Restore

Cisco UE runs on a hardware module that contains storage media separate from that of the router. The NM-CUE has a hard disk, and the AIM-CUE uses onboard compact Flash for storage. The configuration of Cisco UE AA and voice mailincluding the AA prompts, user greetings, spoken names, and voice messagesis stored on this disk or Flash unit. Backing up the router configuration, therefore, is insufficient to preserve the Cisco UE application configuration in the event of a disaster (for example, a hard disk failure) or to restore the system's operational data after an application upgrade.

Note

A Cisco UE software upgrade reformats the hard disk or Flash unit. To preserve the application configuration and voice messages over an upgrade, you must perform a backup before the upgrade and a restore afterwards.

Cisco UE includes a Backup and Restore facility that uses an FTP server as the backup's destination device. You cannot use Flash or other types of media for backup and restore. You might locate the FTP server anywhere in the network, and you do not need to collocate it with the Cisco UE system being backed up. An FTP server login and password provide secure access to the backup and restore operations.

Backups are invoked manually from the Cisco UE system, using either the CLI or GUI interface. The system must be offline during a backup; therefore, no calls might be active in the system when a backup is being done. After a restore operation, you must restart the application.

Although the Cisco UE system has no mechanism to schedule unattended backups, the backup functionality is available through the CLI. Therefore, it is possible for you to develop a script on another server that automatically (for example, based on time of day) logs into the Cisco UE system's CLI and initiates a backup. If you do scheduled backups in this manner, consider the following:

- A backup requires the system to be offline, and taking the system offline disconnects all calls in progress. If the backup is triggered by a script where the warnings about call disconnection are not seen (because of a scripted interface), backups during normal daytime system use disrupt the system's operation. Write your script to initiate backup during a time of day when no calls are expected to be active.
- If any errors occur, ensure that your script includes a notification to an administrator so that the errors can be investigated.

Several aspects important to configuring backups are discussed in the following sections:

- Specifying a directory path for backup
- Backing up multiple Cisco UEs in a network
- Including configuration and data in the backup
- Configuring multiple generations of backups
- Determining backup file sizes

Configuring Interconnection with Other Sites

You might use Cisco IPC Express as a standalone system, or you might interconnect multiple Cisco CME sites. You might also use a centralized Cisco Unity for voice mail. For these types of networks, you have to configure not only the Cisco IPC Express system itself, but also the interconnectivity between the sites.

The topics of site interconnection and examples of the salient aspects of Cisco IPC Express system configuration were covered in other chapters:

- Chapter 7, "Connecting Multiple Cisco CMEs with VoIP"
- Chapter 8, "Integrating Cisco CME with Cisco CallManager"
- Chapter 10, "Cisco IPC Express Integrated Voice Mail"
- Chapter 11, "Cisco CME External Voice Mail Options"

Security Best Practices for Cisco CME

Cisco IPC Express provides integrated IP communications on Cisco IOS routers. Therefore, the same security best practices recommended for all Cisco IOS voice-enabled routers also apply to Cisco CME. In addition, you should implement Cisco CME-specific security practices to provide additional security protection.

This section explains how you can set up the Cisco CME using the CLI to prevent users from intentionally or accidentally gaining system-level control from the GUI as well as local or remote CLI access.

Securing GUI Access

A Cisco IOS router authenticates an administrator CLI login against the enable password only, and the default setting for HTTP access is ip http authentication enable. If the system administrator, customer administrator, or phone user has the same password as the router's enable password, he or she can gain level 15 EXEC privilege access to Cisco IOS via HTTP. A normal IP phone user can then accidentally change the Cisco CME configuration, erase Flash, or reload the router when logging on to this URL:

```
http://cme-ip-address/
```

You should configure the following commands for Cisco CME to use AAA or local authentication to prevent a normal user from gaining access to the enable password and therefore having access to the system administrator page:

```
ip http authentication aaa
or
ip http authentication local
```

Note

Note that authentication, AAA is applied only to the system administrator login. Local authentication, which is clear-text-based, is applied to both the customer administrator and phone user logins.

Using HTTPS for Cisco CME GUI Management

HTTP over SSL (HTTPS) provides Secure Socket Layer (SSL) version 3.0 support for the HTTP 1.1 server and HTTP 1.1 client within Cisco IOS software. SSL provides server authentication, encryption, and message integrity to allow secure HTTP communications. SSL also provides HTTP client authentication. This feature is supported only in Cisco IOS software images that include the SSL feature. Specifically, SSL is supported in the Advanced Security, Advanced IP Services, and Advanced Enterprise Services images. Use the Advanced IP Services or Advanced Enterprise Services Cisco IOS images to get both the Cisco CME and SSL features.

Currently IP phones do not serve as HTTPS clients. If HTTPS is enabled on the Cisco CME router, IP phones still attempt to connect to HTTP using port 80. Because the SSL default port is 443, the phones cannot display local directory and system speed dials. IP phones using HTTP can work with a system configured for SSL by enabling both HTTP and HTTPS, as shown in Example 14-19.

Example 14-19. Enabling HTTP Secure Server Sample Configuration

```
router#show running-config
ip http server
ip http secure-server
ip http secure-port port_number
!if https port is changed from default 443
ip http authentication AAA |TACACS | local
```

Security Best Practices for Cisco UE

You should consider various additional aspects of network security to protect against unauthorized access to Cisco UE. This section covers Cisco UE security best practices related to system access, remote access, and other security parameters applicable to the application environment.

System and Remote Access

Cisco UE hardware does not have external interfaces (physically, there is a Fast Ethernet interface port, but it is disabled in software and unusable). All Cisco UE system access must pass through the host Cisco CME router. Cisco UE CLI access has no login access or password control in addition to that of the router that houses Cisco UE. Therefore, it is imperative that the router's configuration parameters for local access (the console port) and remote access (Telnet) are set according to your security needs.

Local Access

The only local access to a Cisco UE system is via the host Cisco CME router's console interface into the router CLI. You then open a session to the Cisco UE CLI by using the following command:

```
router#service-module service-Engine x/y session
```

Entering this command on the router requires enable mode and, therefore, is protected by the router's enable login and password settings. Although the Cisco UE CLI also has an enable mode, it has no user ID or password capability. Any network administrator who has access to enable mode on the router also has access to the Cisco UE CLI. Access is controlled via the router, so if logging is required, set up the router with AAA/RADIUS monitoring of login access.

GUI access via a browser to Cisco UE is considered remote access, because it is across an IP segment from the router.

Remote AccessTelnet

Routers typically are geographically dispersed in your network and are seldom accessed locally via the console port. Remote access via Telnet across the IP network is much more typical. Use the IP configuration shown in Example 14-29 as a reference for the discussion in this section.

Example 14-29. IP Reference Configuration

```
router#show running-config
interface FastEthernet0/0
 ip address 172.19.153.41 255.255.255.0
 no ip mroute-cache
 duplex auto
 speed auto
!
interface Service-Engine1/0
 ip unnumbered FastEthernet0/0
 service-module ip address 172.19.153.37 255.255.255.0
 service-module ip default-gateway 172.19.153.41
```

Direct Telnet access to the Cisco UE IP address is disabled, as shown in Example 14-30.

Example 14-30. Cisco UE Telnet Access Disabled

```
pc>telnet 172.19.153.37
Trying 172.19.153.37...
```

Configuring and Monitoring Via Network Management Systems Using the Cisco CME AXL/SOAP Interface

You can integrate Cisco CME with network management applications by using the Cisco CME XML Layer (AXL) application programming interface (API). The AXL API provides a mechanism for inserting, retrieving, updating, and removing data from the Cisco CallManager database using an XML SOAP interface. The AXL API allows programmatic access to Cisco CallManager data in XML form instead of using a binary library or a Dynamic Link Library (DLL). The AXL API methods, or requests, are performed using a combination of HTTP and SOAP. The HTTP payload is encapsulated in SOAP, which is essentially an XML remote procedure call protocol. User requests send XML data to the Cisco CallManager server, which returns an AXL response encapsulated in a SOAP message.

Cisco CME extends the AXL/SOAP capabilities by providing XML APIs for monitoring and configuring IP phones and extensions. A Network Management System (NMS) might use the Cisco CME AXL/SOAP APIs to poll the Cisco CME network elements (NEs), including IP phones and extensions. As with the AXL protocol, communication between an NMS and Cisco CME is based on an HTTP data exchange and can be initiated only by polling from the NMS. However, Cisco CME can enable or disable the sending of data, as well as control the polling interval.

Note

AXL/SOAP APIs for NMS configuration and monitoring are supported only by Cisco CME, not by Cisco UE.

The next sections describe the features supported by the Cisco CME AXL/SOAP APIs and a test procedure to check if your Cisco CME is set up properly to respond to the AXL/SOAP queries.

The Cisco CME AXL/SOAP Interface

The Cisco CME AXL/SOAP APIs provide many capabilities for monitoring and configuring IP phones and extensions.

For monitoring, Cisco CME AXL/SOAP APIs support the following:

- Getting static information

  ISgetGlobalGets global information

  ISgetDeviceGets device information

  ISgetExtensionGets extension information

- Getting dynamic information

  ISgetEvtCountsGets the number of events recorded in the buffer

  ISgetDevEvtsGets device events if IP phones are in the register, unregister, or decease state

  ISgetExtEvtsGets extension events (the virtual voice port is up or down)

Monitoring Cisco IPC Express

You might monitor the Cisco IPC Express system with Syslog messages and Simple Network Management Protocol (SNMP) Management Information Base (MIB). You also can monitor call activity information through Syslog messages and Call Detail Records (CDRs).

Monitoring IP Phones Using Cisco CME Syslog Messages

Cisco CME 3.0 introduced type 6 Syslog messages, as shown in Example 14-37, for IP phone registration and deregistration events. These Syslog messages are useful for a central NMS to manage Cisco CME systems and IP phones.

Example 14-37. Type 6 Syslog Messages

```
%IPPHONE-6-REG_ALARM
%IPPHONE-6-REGISTER
%IPPHONE-6-REGISTER_NEW
%IPPHONE-6-UNREGISTER_ABNORMAL
%IPPHONE-6-REGISTER_NORMAL

Example Message:
 %IPPHONE-6-REGISTER_NEW: ephone-3:SEP003094C38724 IP:1.4.170.6 Socket:1
  DeviceType:Phone has registered.
```

The IPPHONE-6-REGISTER_NEW message shown in Example 14-37 indicates that a phone has registered and that it is not part of the explicit router configuration. In other words, the ephone configuration has not yet been created. Cisco CME allows unconfigured phones to register to make provisioning of the Cisco CME system more convenient. By default, phones designated as new are not assigned phone lines; therefore, they cannot make calls until they are configured into the system.

Enable the Cisco IOS logging capability to log all the Syslog events into the buffer on the Cisco CME router, or send the Syslog messages to a Syslog server for offline management, as shown in Example 14-38.

Example 14-38. Enabling Syslogging

```
telephony-service#(config)#service timestamps log datetime msec localtime
telephony-service #(config)#aaa new-model
telephony-service #(config)#aaa authentication login default none
telephony-service #(config)#aaa accounting connection H.323 start-stop radius
telephony-service #(config)#gw-accounting syslog
telephony-service #(config)#logging 10.10.10.1
!!! 10.10.10.1 is the ip address of syslog server, multiple servers might also be
  specified
```

To synchronize your Cisco CME system to an external NTP server, use the following:

```
ntp server ip-address
 !!! ip address - IP address of the time server providing the clock
  synchronization
```

If there is no external NTP time source, use the internal router clock as the time source:

ntp master

To ensure that the time stamps are correct, set the router clock to the correct time:

Managing Cisco IPC Express Systems by Managed Services and Enterprises

As described in earlier chapters, Cisco IPC Express is ideal if you have data connectivity requirements and also need IP telephony in the office. SPs normally deploy Cisco IPC Express systems as one of the following:

- Standalone, single-site managed services
- Large-scale, multisite managed services

A managed-services solution with Cisco CME offers two opportunities for value-added services:

- The customer premises equipment (CPE) router
- Network management support

SPs offer their customers the Cisco IPC Express systems at the end customer's site. They also install, set up, maintain, and manage the systems.

Most of the NMSs used by SPs to deploy Cisco IPC Express in a managed-services model also apply to enterprise networks. The difference between a managed-services model and an enterprise model is who offers, owns, and manages the core network.

Note that this chapter covers only management capabilities for Cisco IPC Express systems, not for the larger IP telephony solutions and products offered by Cisco in general. The next sections describe how you can manage standalone or multisite Cisco IPC Express systems. They also cover some general information on the typical Cisco Voice Network Management Solutions that are applicable to Cisco IPC Express.

Managing a Standalone Cisco IPC Express System

Figure 14-43 shows a deployment in which a single Cisco IPC Express system in a branch office connects to a SP's VoIP network. All voice and data traffic can be routed over the SP's network, or calls can be routed via the PSTN if the destination (called party) cannot be reached via the SP's IP network.

Figure 14-43. Managing a Standalone Cisco IPC Express System

[View full size image]

Cisco Voice Network Management Solutions

This section describes what Cisco network management applications you can use to manage Cisco IPC Express systems. Some of these solutions might not manage Cisco IPC Express features directly but might facilitate the management of the underlying voice network.

Cisco offers management tools that dramatically reduce the cost of operating a network. By providing tools that reduce human intervention in the initial deployment of network hardware and ongoing maintenance, network management helps decrease human error and increases the total number of devices that a single person can manage. Cisco provides network management tools to help SPs and enterprise customers manage their data and voice networks. The network management options are via either the CLI for standalone sites or automated NMSs for large networks.

The NMS solution is an automated service deployment option that uses integrated network management devices to provide fault, configuration, accounting, performance, and security (FCAPS) management support. Because a Cisco IPC Express system runs as part of the Cisco IOS voice-enabled router, all the existing Cisco Voice Network Management systems are applicable, including the following:

- CiscoWorks QoS Policy Manager (QPM)
- CiscoWorks Resource Manager Essentials (RME)
- Internetwork Performance Monitor
- Cisco Voice Manager and Telemate

The next sections describe each of these systems in more detail.

CiscoWorks QoS Policy Manager

CiscoWorks QoS Policy Manager (QPM) is a management tool for QoS policy creation, deployment, and monitoring. It provides rules-based policy guidance plus templates (designed by Cisco or customized by users) to streamline QoS management. It improves efficiencies by administering QoS for IP telephony across the entire network using step-by-step wizards and templates based on Cisco design recommendations. CiscoWorks QPM enforces network-wide services such as low latency, as well as fast response for VoIP, video, and other applications, such as SAP, Oracle, and PeopleSoft.

CiscoWorks Resource Manager Essentials

CiscoWorks Resource Manager Essentials (RME) is an infrastructure tool for full-service branch management. It helps streamline a network's daily operations. The CiscoWorks RME browser interface allows easy access to information critical to network uptime and simplifies time-consuming administrative tasks. The Management Connection feature in CiscoWorks RME adds web integration of other management tools from Cisco and partner companies. Therefore, you can use these tools and applications to create a seamless central point of network administration. CiscoWorks RME includes the following components:

- Inventory Manager Manages inventory of the devices
- Change Audit Tracks changes made to each managed device

Managing Cisco IPC Express with Cisco Partner Applications

In addition to the Cisco management solutions discussed in the previous sections, some Cisco partners offer management solutions. This section describes two of these solutions:

- NetIQ Vivinet Manager
- Stonevoice

NetIQ Vivinet Manager

NetIQ's Vivinet Manager allows you to gain access to Cisco CME data, and then analyze and manage Cisco CME Systems. With NetIQ Vivinet Manager for Cisco CME, you gain easy access to a new set of tools you can leverage to gather a wide range of diagnostic and management data, which can help prevent outages and keep things running smoothly.

NetIQ Vivinet Manager for Cisco CME is an add-on module to NetIQ Vivinet Manager version 2.1. Equipped with Cisco CME AXL/SOAP API support, you can use Knowledge Scripts included in Vivinet Manager for Cisco CME to create jobs that monitor the health, availability, and performance of key devices. These scripts allow you to monitor and manage crucial device properties at a depth unparalleled by any other solution. You can configure each Knowledge Script to send an alert, collect data for reporting, and perform automated problem management when an event occurs.

The Vivinet Manager Knowledge Scripts let you monitor phone status (registered, unregistered, and deceased), reset IP phones, specify key phones, monitor for duplicate extensions, and show inventory information for phones attached to Cisco CME systems. The following are the supported Knowledge Scripts for the Cisco CME module:

- CiscoCME_Device_Reset Resets Cisco CME IP phones for reasons such as troubleshooting or picking up new default firmware. Use this script in conjunction with CiscoCME_Device_Status to ensure that the selected phones have upgraded successfully.
- CiscoCME_Device_Status Monitors the status of key Cisco CME phones.
- CiscoCME_Extension_Check Monitors for duplicate phone extension numbers. This script looks for all phones configured in Cisco CME, regardless of whether they are registered.
- CiscoCME_Phone_Inventory Generates an inventory of the phone details for phones that are attached to Cisco CME.
- CiscoCME_Set_Key_Phones Designates one or more "key" phones. After you designate key phones, you can choose to monitor only key phones.

The following features are provided by NetIQ Vivinet Manager for Cisco CME:

- It discovers Cisco CMEs with CME version and device information.
- It provides Knowledge Scripts for day-to-day and diagnostic monitoring.

Summary

This chapter covered how many of the different features of a Cisco IPC Express system are configured, including Cisco CME phone and call processing features, as well as Cisco UE AA and voice mail features.

It also gave an overview of the network management applications available for Cisco voice (IP telephony) solutions for small and medium businesses and enterprise branch offices. It covered managing Cisco IPC Express when deployed as a managed service CPE by a SP. This chapter also covered details on using the Cisco partner applications NetIQ Vivinet Manager with Cisco CME (interfacing via Cisco CME's SOAP AXL APIs) and Stonevoice.

Chapter 15. Cisco IPC Express System Configuration Example

This chapter covers the following topics:

- Configuring a two-site sample Cisco IP Communications (IPC) Express system
- Planning and offline staging
- Basic router setup
- Initial Cisco CallManager Express (CME) system setup
- Configuring extensions and phones
- Configuring the Public Switched Telephone Network (PSTN) interface
- Configuring Cisco Unity Express (UE) automated attendant (AA) and voice mail
- Configuring Cisco CME call processing features
- Interconnecting multiple Cisco IPC Express systems
- Listing of the Site A and Site B Cisco CME and Cisco UE configurations built in this chapter

This chapter provides a step-by-step configuration guide to help you set up your Cisco IP Communications (IPC) Express system. System configuration and setup topics, features, and details were discussed in depth in Chapter 13, "Cisco IPC Express General Administration and Initial System Setup," and Chapter 14, "Configuring and Managing Cisco IPC Express Systems." This chapter helps you understand that information by walking you through configuring a newly ordered system to create a working system for your office. Not every feature is covered again in this chapter. Instead, it focuses on the steps you should follow and draws on the previous chapters for many of the details.

This chapter uses a two-site sample network, as shown in Figure 15-1. This chapter walks through the setup and configuration of Site A, the primary site. Site B has a very similar basic configuration and is used to illustrate what extra configuration to do (as soon as you have the standalone system running with Site A) to connect two Cisco IPC Express sites.

Figure 15-1. Cisco IPC Express Network with Two Sites

[View full size image]

Step 1: Planning and Offline Staging

Before building a configuration for a Cisco IPC Express system, you should note the parameters of the configuration you want to put in place. This includes such information as the names of the employees in your office, the extension numbering scheme you want to use, the Public Switched Telephone Network (PSTN) numbers your office has assigned, where PSTN calls should be routed, what type of phone each employee and room will have, what the IP addressing of your office is, and where the Trivial File Transfer Protocol (TFTP) server is located.

You can proceed from here in two ways, depending on whether you have the equipment ready in your office or lab:

- If you have already ordered the equipment and it has been delivered, or if you are reusing existing routers and IP phones, move on to Step 2, Basic Router Setup.
- If you have no equipment readily available, but you want to start building a configuration for staging purposes, read about offline staging in this section before moving on. The rest of this chapter cannot be executed without access to the equipment.

A Cisco CME Installation Configuration Tool (ICT) is an offline HTML tool available as a shareware application provided for Cisco Partners and Resellers. You can download the tool from the Cisco.com software center for Cisco CME, and use it to set up the basic telephony-service, IP phone, and voice mail configuration for all Cisco CME supported platforms.

You fill in basic fields about the system's desired configuration and scan in the phones' Media Access Control (MAC) addresses (if they are available). The tool output provides the router configuration (command-line interface [CLI]) as well as the Cisco UE configuration, which you can cut and paste into the equipment when that arrives.

Bring up a Microsoft Internet Explorer (IE) browser with the expresso.htm file. Note that expresso.htm is the only file that serves as an entrance to the tool. Figure 15-2 shows this tool's main screen, where you enter the number of phones, the IP addressing, whether you have voice mail, and, if so, the pilot numbers for AA and voice mail.

Figure 15-2. Cisco CME ICT System Parameters

[View full size image]

Click the Phone Parameters tab on the left panel to go to the next screen, shown in Figure 15-3, where you can enter

Step 2: Basic Router Setup

As soon as you have the equipment on hand, the configuration of the real system can start. You can do this either by copying and pasting the staging configuration you built in the preceding section or by working through the following sections to set up the basic router and then the Cisco CME and Cisco UE parameters. Even if you use the staging configuration, it is still recommended that you scan the steps in this chapter. This will help you make sure that the parameters in the staged configuration are, in fact, the values you want for your real system configuration. It also helps you make any adjustments along the way to tailor the system for your use.

Installing Hardware and Software

Cisco routers and Cisco UE ship from the factory preinstalled with the hardware and software you ordered. Unless you have to upgrade the software, there is no need to do an installation when you unpack your equipment. If you have to make changes from what was ordered, or if you're reusing older equipment, power down the router, insert the hardware components, and power the router back up. Refer to Cisco.com for instructions for all types of hardware.

Follow the software installation instructions for Cisco CME and Cisco UE provided in Chapter 13. Ensure that you have an IP Voice or greater image running on your router. You should install the releases of Cisco CME and Cisco UE that have the feature complement you're interested in. Install at least Cisco CME 3.2 (12.3.11T) or Cisco UE release 2.0 to get the features discussed in this chapter.

Configuring Router IP Addressing

Connect to your router's console port, and enter an IP address for the Ethernet interface so that you can connect the router to your network:

```
interface FastEthernet0/0
 ip address 10.1.235.1 255.255.0.0
```

Also, set your router's host name to a descriptive string. For Site A, the name is cme-3725:

```
hostname cme-3725
```

Setting the Router Clock

Ensure that the clock is set correctly on your router, for example:

```
cme-3725#clock set 10:00:00 19 Aug 2004
```

You might also set the router clock using Network Time Protocol (NTP). The details are given in the section "Configuring NTP."

Setting Up the LAN Switch

You can use either an external LAN switch or an internal EtherSwitch housed inside the router chassis. You should define virtual LAN (VLAN) for both voice and data traffic.

External LAN Switch

For an external LAN switch, to enable separate VLANs for voice and data on a single router port, configure a trunk between the Cisco CME router and the LAN switch. The Cisco Catalyst 3550 and 3560 support autodetection of the VLAN type for IP phones (voice VLAN) and PCs (data VLAN). VLAN configuration varies between LAN switch types. Consult the appropriate documentation for the LAN switch model you are using.

Step 3: Initial Cisco CME System Setup

You have now set up the basic routing and LAN switching. The next task to accomplish is to start setting up the basic Cisco CME configuration. You can accomplish this in several ways:

- Use the Cisco CME Setup Utility
- Copy and paste the configuration, or segments thereof, from the staging configuration (if you used that tool)
- Configure the commands individually on Cisco CME

In the next section, you use the Cisco CME Setup Utility to do the basic Cisco CME configuration. If you are not using this utility, check the resulting CLI from the sections following the discussion of the Setup Utility, and compare that to the CLI that exists in your configuration to ensure that all the different parameters are set to the desired values.

Refer to the related sections in Chapter 13 for details on the configuration for different features.

Running the Cisco CME Setup Utility

Example 15-2 provides a log of running the Cisco CME Setup Utility (discussed in Chapter 13) to build the Site A basic configuration.

Example 15-2. Cisco CME Setup Utility for Site A

```
cme-3725(config)#telephony-service setup
 --- Cisco IOS Telephony Services Setup ---
Do you want to setup DHCP service for your IP Phones? [yes/no]: yes
Configuring DHCP Pool for Cisco IOS Telephony Services :
  IP network for telephony-service DHCP Pool:20.1.1.0
  Subnet mask for DHCP network :255.255.255.0
  TFTP Server IP address (Option 150) :20.1.1.100
  Default Router for DHCP Pool :20.1.1.100
Do you want to start telephony-service setup? [yes/no]: yes
Configuring Cisco IOS Telephony Services :
  Enter the IP source address for Cisco IOS Telephony Services :20.1.1.100
  Enter the Skinny Port for Cisco IOS Telephony Services : [2000]:
  How many IP phones do you want to configure : [0]:_5
  Do you want dual-line extensions assigned to phones? [yes/no]: yes
  What Language do you want on IP phones [0]: 0
  Which Call Progress tone set do you want on IP phones [0]:
  What is the first extension number you want to configure : 2001
Do you have Direct-Inward-Dial service for all your phones? [yes/no]: no
Do you want to forward calls to a voice message service? [yes/no]: yes
  Enter extension or pilot number of the voice message service: 2105
  Call forward No Answer Timeout : [18]: 10
Do you wish to change any of the above information? [yes/no]: no
 ---- Setup completed config ---
cme-3725(config)#
2w3d: %LINK-3-UPDOWN: Interface ephone_dsp DN 1.1, changed state to up
2w3d: %LINK-3-UPDOWN: Interface ephone_dsp DN 1.2, changed state to up
2w3d: %LINK-3-UPDOWN: Interface ephone_dsp DN 2.1, changed state to up
2w3d: %LINK-3-UPDOWN: Interface ephone_dsp DN 2.2, changed state to up
2w3d: %LINK-3-UPDOWN: Interface ephone_dsp DN 3.1, changed state to up
2w3d: %LINK-3-UPDOWN: Interface ephone_dsp DN 3.2, changed state to up
2w3d: %LINK-3-UPDOWN: Interface ephone_dsp DN 4.1, changed state to up
```

Step 4: Configuring Extensions and Phones

On Cisco CME you normally configure DNs (ephone-dns) before configuring the phones (ephones). This is different from Cisco CallManager, where phones must register before you configure extensions. On Cisco CME, you can configure extensions regardless of whether the phones are registered. The following sections walk you through setting up phones and extensions on this chapter's sample system.

Defining Extensions

The Cisco CME Setup Utility has already created basic DN and phone definitions for the number of phones you specified. For the Site A setup, five phones were specified.

One of the Setup Utility questions that inquires about dual-line phones is, "Do you want dual-line extensions assigned to phones?" For Site A, the reply was yes (refer to Example 15-2). A dual-line configuration is needed for features such as call transfer, call waiting, and call conferencing. A dual-line configuration uses one line with two channels so that a second call to the same line can be put in call waiting mode, or a call transfer or conference call can be initiated using the second channel.

You cannot change a nondual-line ephone-dn to dual-line mode. You have to insert the ephone-dn with this mode. If you chose dual-line mode in the Setup Utility, as in Example 15-2, you're all set. If not, you may have to delete the ephone-dns created by the utility. (Use the command no ephone-dn x, where x is the number of the ephone-dns.) Reenter them in dual-line mode using the ephone-dn x dual-line command.

Example 15-7 shows the first two of the five ephone-dns created by the utility log shown in Example 15-2. Because the replies in the utility also specified that a voice mail system existed and that the pilot number was 2105, the call forward CLI has already been entered for the ephone-dns.

Example 15-7. Ephone-dns Created by the Setup Utility

```
cme-3725#show running-config
ephone-dn 1 dual-line
 number 2001
 call-forward busy 2105
 call-forward noan 2105 timeout 10
!
ephone-dn 2 dual-line
 number 2002
 call-forward busy 2105
 call-forward noan 2105 timeout 10
```

You can create additional ephone-dns using the CLI, or you can log into the GUI at this point and add extensions via the Configuration > Extensions screen.

Many types of DNs and different ephone-dn features exist. The configuration done so far in this section provides only the basic extension (DN) configuration necessary to make calls, but you have not yet configured the phones. The following section shows you how to complete some basic phone configuration.

Assigning Extensions to IP Phones

The Cisco CME Setup Utility shown in Example 15-2 has already entered ephone definitions into the configuration. They show up as a single line only in the CLI, as shown here, until the phones are powered, connected, and registered with Cisco CME:

```
ephone 1
ephone 2
```

Step 5: Configuring the PSTN Interface

The next step in Cisco CME system configuration is to route PSTN calls into your office to the extensions on the IP phones, and to allow IP phones to make outgoing PSTN calls. The following sections cover voice port and PSTN trunk configurations necessary to route PSTN calls to IP phones.

Configuring Voice Ports

Chapter 6, "Cisco CME PSTN Connectivity Options," explained the various hardware options you have on Cisco CME for connecting to the PSTN and also provided several configuration examples for different trunk types. The following sections build examples of basic PSTN connectivity for both an analog Foreign Exchange Office (FXO) and a digital T1 Primary Rate Interface (PRI) trunk. It is likely that your office will use one or the other, but not both.

If analog hardware is present in your Cisco CME system, ports show up automatically in the router configuration as follows:

```
voice-port 2/0/0
voice-port 2/0/1
```

To add caller ID to analog FXS or FXO ports, use the caller-id command, as shown in Example 15-13.

Example 15-13. Configuring Caller ID on FXS or FXO Ports

```
cme-3725#show running-config
voice-port 2/0/0
 caller-id enable
!
voice-port 2/0/1
 caller-id enable
```

Digital ports do not show up in the configuration simply because the hardware is present in your router. All you see by default is the controller statement alerting you that T1 or E1 port hardware is present. On the newer controller cards that allow software configuration for either T1 or E1 operation on the same hardware, the controller doesn't show up in the configuration until you configure the port type to be either T1 or E1 operation. Use the card type command to accomplish this.

Example 15-14 shows the configuration for a T1 PRI trunk type with the 5ESS switch type. Which switch type you should use depends on the central office you connect to and varies between geographic locations, as well as for T1 compared to E1 ports. The voice port, 0/0:23 in Example 15-13, is created automatically by the pri-group configuration, as is the D channel interface (interface Serial0/0:23 in Example 15-13). If you have T1 Channel Associated Signaling (CAS) or E1 R2 connectivity to the PSTN, you use the ds0-group command instead of the pri-group command.

Example 15-14. Configuring a T1 PRI PSTN Trunk

```
cme-3725#show running-config
network-clock-participate wic 0
network-clock-participate aim 0
isdn switch-type primary-5ess
!
controller T1 0/0
 pri-group timeslots 1-24
!
interface Serial0/0:23
 no ip address
 isdn switch-type primary-5ess
 isdn incoming-voice voice
```

Step 6: Configuring Cisco UE AA and Voice Mail

If your system does not have Cisco UE installed, you can skip this section and proceed to Step 7. The following sections step you through setting up the basic Cisco UE configuration necessary to use the AA and voice mail on your system.

Setting Up the Router for Cisco UE

Before you can configure and use Cisco UE applications such as AA and voice mail, you must set up the following basic IP connectivity parameters to ensure that the Cisco UE software can communicate with its environment:

- IP addressing
- Call routing to Cisco UE
- H.323-to-Session Initiation Protocol (SIP) call routing
- Message Waiting Indicator (MWI)

IP Addressing

The Cisco UE hardware module must be configured with an IP address on the router. This setup is covered in Chapter 13. Following the information provided there, the resulting configuration for the Cisco UE module for Site A is shown in Example 15-18.

Example 15-18. Service-Engine Interface for Cisco UE

```
cme-3725#show running-config
interface Service-Engine1/0
 ip unnumbered FastEthernet0/0
 service-module ip address 10.1.235.128 255.255.0.0
 service-module ip default-gateway 10.1.235.1
ip route 10.1.235.128 255.255.255.255 Service-Engine1/0
```

You can show the status of the Cisco UE module by using the command shown in Example 15-19. You can also ping the IP interface (10.1.229.128 in this example) to ensure that IP connectivity is established.

Example 15-19. Showing the Status of the Cisco UE Module

```
cme-3725#service-module service-engine 1/0 status
Service Module is Cisco Service-Engine1/0
Service Module supports session via TTY line 33
Service Module is in Steady state
Getting status from the Service Module, please wait..
cisco service engine 1.1
```

Call Routing to Cisco UE

Cisco CME and Cisco UE communicate using a SIP interface. Therefore, Cisco CME uses SIP dial peers to determine which calls must be routed to Cisco UE. The AA pilot number for Site A is 2100, and the voice mail pilot is 2105. Example 15-20 shows the SIP dial peers necessary to route calls to these pilot numbers from Cisco CME to Cisco UE.

Step 7: Configuring Cisco CME Call Processing Features

Your Cisco IPC Express system is fully operational after the completion of Step 6 for all basic features. However, you probably want to configure numerous additional Cisco CME call processing features to better tailor the system to your business needs. The next sections describe configuring the following:

- Phone and user features
- System features such as music on hold (MOH) and paging
- Call forward, transfer, and conference
- Phone applications

Configuring Phone and User Features

In this section, you configure several phone and user-related features:

- Caller ID name display
- Phone name display
- Phone button label customization
- Shared lines
- Hunt group
- Local directory
- Speed dial
- Local speed dial
- Personal speed dial
- Localization
- Autoline selection
-

Step 8: Interconnecting Multiple Cisco IPC Express Systems

In the preceding seven steps, you set up the entire Cisco IPC Express system at Site A with examples of the most commonly deployed features you will require for your office. Although the Site B configuration was not discussed step by step, its configuration is very similar to that for Site A, with extensions starting at 3001 instead of 2001.

You cannot yet make calls between Sites A and B, because there is no dial plan to route calls between the sites. To achieve this, first ensure IP routing between the sites. In the sample configuration being built in this chapter, Site A has an IP address of 10.1.235.1 (with a netmask of 255.255.0.0), and Site B has an IP address of 10.1.229.1 (with a netmask of 255.255.0.0), so these systems can easily reach each other. If your sites' IP addressing is more sophisticated, do the necessary configuration to achieve IP routing between your sites. Ensure that you see routes between the IP addresses of your sites with a show ip route command on each site's router. You should also be able to ping one site from the other.

Interconnecting Sites Via H.323

As soon as you have IP connectivity between the sites, the next step is to add dial peers to route calls between the sites. From Site A, if someone dials an extension that starts with 3, the call must be routed to Site B. Similarly, if someone at Site B dials an extension starting with 2, the call must be routed to Site A. If your dialing plan is less uniform than the sample network in this chapter, you may need multiple dial peers to route all calls. Also, if only one site has PSTN access, and DID numbers for both the 2xxx and 3xxx ranges arrive on one PSTN trunk, more dial peers are needed to route all calls correctly.

For the sample network, the dial peers to route calls between the sites are shown in Example 15-49.

Example 15-49. Sites A and B Dial Peers

```
! Site A (2xxx extension) dial-peers to direct calls to Site B (3xxx extensions)
cme-3725#show running-config
dial-peer voice 3000 voip
 destination-pattern 3...
 session target ipv4:10.1.229.1
 dtmf-relay h245-alphanumeric
 codec g711ulaw
 no vad
!
! Site B (3xxx extension) dial-peers to direct calls to Site A (2xxx extensions)
cme-2691#show running-config
dial-peer voice 2000 voip
 destination-pattern 2...
 session target ipv4:10.1.235.1
 dtmf-relay h245-alphanumeric
 codec g711ulaw
 no vad
```

At this point, the sites can call each other by simply dialing the extensions of the IP phones. One more configuration must be added to ensure that you can transfer calls between the sites. Add the transfer-patterns shown in Example 15-50 to both sites' configurations.

Example 15-50. Sites A and B Transfer Patterns

```
cme-3725#show running-config
telephony-service
 transfer-system full-consult
 transfer-pattern 3...
 transfer-pattern 2...
```

Sample System Configurations

This section provides the full configurations of Site A and Site B built during the steps in this chapter.

Site A Cisco CME Router Configurations

Site A is a Cisco 3725 router with extensions in the 2xxx range. Figure 15-1, at the beginning of the chapter, summarized the site layout. This section provides the Cisco CME configuration.

The show version Output

Example 15-53 provides the show version output for the Site A Cisco CME router.

Example 15-53. show version Output for the Site A Router

```
cme-3745#show version
Cisco IOS Software, 3700 Software (C3745-IPVOICE-M), Version 12.3(11)T2, RELEASE
  SOFTWARE (fc1)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2004 by Cisco Systems, Inc.
Compiled Fri 29-Oct-04 06:38 by cmong
ROM: System Bootstrap, Version 12.2(8r)T2, RELEASE SOFTWARE (fc1)
cme32-3745 uptime is 2 weeks, 2 days, 4 hours, 39 minutes
System returned to ROM by reload at 15:40:35 PST Wed Dec 8 2004
System restarted at 14:05:42 PST Wed Dec 8 2004
System image file is "flash:c3745-ipvoice-mz.123-11.T2"

Cisco 3725 (R7000) processor (revision 0.1) with 111616K/19456K bytes of memory.
Processor board ID JAB0606800M
R7000 CPU at 240MHz, Implementation 39, Rev 3.3, 256KB L2 Cache
2 FastEthernet interfaces
31 Serial interfaces
2 terminal lines
2 Channelized T1/PRI ports
1 ATM AIM
2 Voice FXS interfaces
2 cisco service engine(s)
DRAM configuration is 64 bits wide with parity disabled.
55K bytes of NVRAM.
31360K bytes of ATA System CompactFlash (Read/Write)
```

The show running-config Output

Example 15-54 provides the show running-config output for the Site A Cisco CME router.

Example 15-54. show running-config Output for the Site A Router

```
cme-3725#show running-config
Building configuration...
Current configuration : 8014 bytes
! Last configuration change at 22:13:14 PST Sat Oct 2 2004
! NVRAM config last updated at 21:35:27 PST Sat Oct 2 2004
!
version 12.3
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname cme-3725
!
boot-start-marker
```

Summary

This chapter walked you step-by-step through a sample setup of a two-site network. The system at Site A was configured from the start to a fully operational system. Additionally, a number of key Cisco CME call processing and phone features were configured on Site A. Site B's configuration was not explored in such great detail. It was used primarily to illustrate the configuration necessary to connect two sites and route calls between them. For the most part, Site B's configuration mirrors that of Site A but is less sophisticated in Cisco CME features containing just the basics to get the system operational.

After working through this chapter, you should be able to configure your own system, and customize it to your or your customer's needs. The content in this chapter solidifies all the feature and management explanations contained in the chapters in Part III. Part IV, starting with the next chapter, is dedicated to troubleshooting and maintaining a Cisco IPC Express system.

Part IV: Maintenance and Troubleshooting

Chapter 16: Troubleshooting Basic Cisco IPC Express Features

Chapter 17: Troubleshooting Advanced Cisco CME Features

Chapter 18: Troubleshooting Cisco CME Network Integration

Chapter 19: Troubleshooting Cisco UE System Features

Chapter 20: Troubleshooting the Cisco UE Automated Attendant

Chapter 21: Troubleshooting Cisco UE Voice Mail Features

Chapter 16. Troubleshooting Basic Cisco IPC Express Features

This chapter covers the following topics:

- Understanding the phone bootup sequence
- Understanding the messages on the phone display
- Debugging virtual LAN (VLAN), Dynamic Host Configuration Protocol (DHCP), Trivial File Transfer Protocol (TFTP), and other registration issues
- Upgrading phone loads with signed and unsigned loads
- Troubleshooting basic call setup and teardown issues
- Troubleshooting Cisco IP Communications (IPC) Express graphical user interface (GUI) issues

Parts II and III of this book covered Cisco IP Communications (IPC) Express features, applications, and management. In Part IV the focus turns to troubleshooting tools and techniques for Cisco IPC Express. The first three chapters of Part IV focus on Cisco CallManager Express (CME), and the last three concentrate on Cisco Unity Express (UE).

Cisco CME provides a range of troubleshooting tools to isolate the cause of problems that might occur. Issues can range from simple situations, such as getting a phone registered to Cisco CME to more complex scenarios, such as overlay-dns and huntgroup configurations not operating correctly. Quite often you can correct problems with small configuration changes or by adjusting the system's operation to be closer to the expected behavior.

Troubleshooting a Cisco CME system requires an understanding of the following:

- The Cisco CME system architecture and its integration with the Cisco IOS voice architecture. This was covered in Chapter 3, "Cisco IPC Express Architecture Overview."
- The phone registration process.
- Cisco CME debug commands.

This chapter explores some basic troubleshooting topics, such as phone bootup and registration, VLAN configurations, Trivial File Transfer Protocol (TFTP) and call setup and teardown, and GUI issues. Subsequent chapters take a closer look at specific features and applications of the system.

Troubleshooting Phone Registration

This section examines the phone's behavior during registration, describes common problems encountered during registration, and illustrates the tools you use to troubleshoot these problems.

The IP phone registration process is fundamental to solving many problems encountered during initial Cisco CME setup. A good understanding of the phone bootup and registration process, phone display messages, and relevant debugs can help you troubleshoot most of these types of problems.

Understanding the Phone Bootup Sequence

When powered on, a phone goes through various states before it attempts to register with a Cisco CME. This section provides an overview in which you learn the correct behavior of an IP phone when it boots up and registers with Cisco CME. The next section provides a step-by-step explanation of the states the phone goes through and what the messages on the phone display mean. The section "Debugging VLAN, DHCP, TFTP, and Registration Issues" discusses problems and issues that may arise when phones don't register correctly.

The following steps summarize the phone bootup sequence; the rest of this section describes the process in more detail:

1. The phone sends out Cisco Discovery Protocol (CDP) messages to discover the voice virtual LAN (VLAN) information from the switch.

1. If the LAN switch is configured with voice VLAN information, the phone receives its voice VLAN information from the switch.

1. The IP phone broadcasts a Dynamic Host Configuration Protocol (DHCP) request to the LAN.

1. The DHCP server responds to the DHCP request with IP address, default gateway, and TFTP server information.

1. The IP phone requests a file named SEPxxxxyyyyzzzz.cnf.xml, where xxxxyyyyzzzz denotes the IP phone's Media Access Control (MAC) address from the TFTP server.

1. The TFTP server sends the requested file to the IP phone. The file also contains the IP address of the Cisco CME router.

1. The IP phone sends a registration request to the Cisco CME router.

1. The Cisco CME router sends the phone configuration to the IP phone.

When powered on, an IP phone issues a CDP request to discover its VLAN information from the LAN switch. It then broadcasts a DHCP request to obtain an IP address. An IP phone's default operation is to broadcast for DHCP. If you want to, you can manually disable this operation.

You can configure a DHCP server on the LAN or across the WAN at another site. The DHCP server responds to the DHCP request with the IP address, default gateway, and TFTP server information if configured. It is recommended that the TFTP server be the Cisco CME router. Upon receiving its IP information, the IP phone downloads a file named SEPxxxxyyyyzzzz.cnf.xml, where xxxxyyyyzzzz denotes the IP phone's MAC address from the TFTP server. Some of the important information contained in this file includes

- IP address of the Cisco CME router
-

Understanding SCCP and Call Flow Debugging

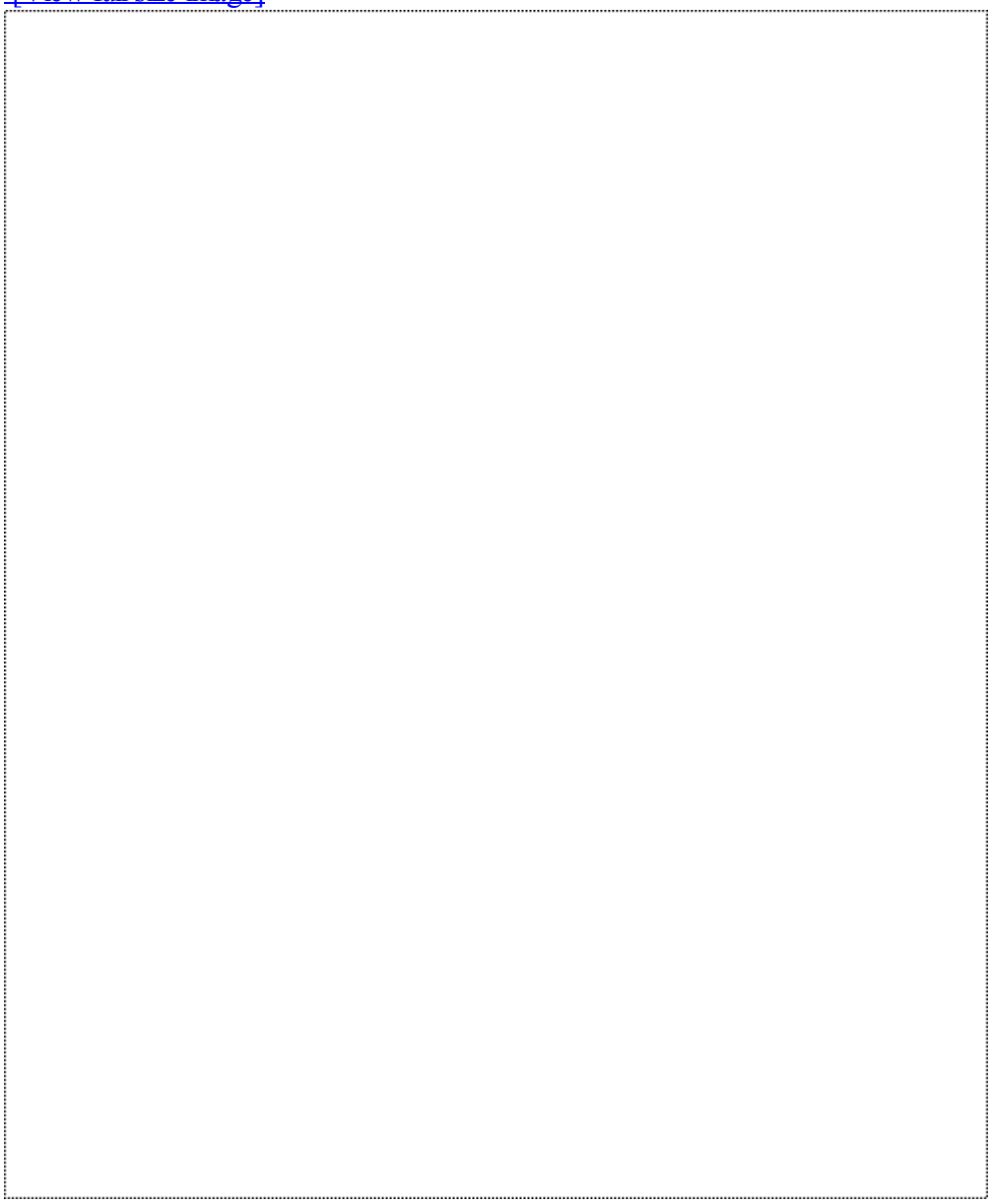
This section describes the SCCP and H.323 messages during call setup and debugging problems related to call setup. Understanding call flow is very useful in troubleshooting call control-related problems.

SCCP to SCCP Call

Figure 16-2 shows a sample exchange of messages between two SCCP clients (IP phones) during call setup. The messages are self-explanatory. You can view the SCCP call control messages by turning on the debug ephone detail command.

Figure 16-2. SCCP to SCCP Call Setup Messages

[View full size image]



Note

The output of debug ephone detail is quite verbose and shouldn't be turned on for a Cisco CME in a production environment with large call volumes. Doing so can degrade the router's performance considerably. You can turn on the debugs for specific phones by adding the mac-address option to the debug, thus limiting the debug output.

Troubleshooting the Cisco IPC Express GUI

The GUI is a very useful tool for configuring and maintaining both Cisco CME and Cisco UE, especially for a system with a higher number of ephones and ephone-dns configured. The GUI is a shared interface for Cisco IPC Express and, therefore, contains elements for both Cisco CME and Cisco UE.

Common GUI Problems and Solutions

This section explains some of the basic problems that may come up while using GUI:

- The browser doesn't start
- Lack of IP connectivity
- Incorrect browser version and wrong browser settings
- The HTTP server isn't enabled on the router
- GUI files are unavailable in the router's flash

You also learn how to determine what the problem is.

The Browser Doesn't Start

When the browser doesn't start, take the following approach to narrow down the problem:

Step 1. Check IP connectivity between your PC and the Cisco CME router.

Step 2. Check the browser and revision. Only Internet Explorer (IE) 6.0 with JScript 5.6 or later is supported.

Step 3. Check the IOS configuration. You should see the ip http server and ip http path flash: statements in the router configuration.

Step 4. Ensure that all the required Cisco CME GUI files are present in the router's flash. Missing a crucial GUI file, or having the wrong version of the file, causes the browser not to launch properly.

Step 5. Determine if the problem is with the browser. Use the same browser to access Internet web pages to see if the problem persists.

Step 6. There may be conflicts with other Java installations running on the PC. Disable other Java applications to see if that improves the situation.

IP Connectivity

To check IP connectivity between your PC and the Cisco CME router, open a DOS command window on your PC and ping the IP address of the Cisco CME router. The PC should be able to ping the router with a 100% success rate.

The HTTP Server Isn't Enabled on the Router

Summary

This chapter covered some of the initial system setup problems you may encounter, such as the registration of IP phones with Cisco CME and GUI connectivity operation.

You learned various ways of identifying phone registration problems and the configuration changes to solve them. The phone display is one of the key tools used to isolate problems with phone registration and operation. The areas of VLAN, DHCP, and TFTP configurations were examined. You saw how these settings may affect phone registration issues.

Basic call flow debugging was covered. You learned how to trace a call setup from off-hook to dialing, and media establishment to call disconnect. You can use this knowledge if you have to troubleshoot a call setup or establishment problem.

This chapter also discussed the basic techniques of troubleshooting GUI issues when administering Cisco IPC Express. It showed the steps required to narrow down the problem and then use the available commands on Cisco CME and Cisco UE, such as debug and trace, to obtain more detailed information to determine the problem. This chapter covered the basic setup options for ensuring that the GUI works properly and the browser settings are correct.

The following chapters look at more advanced troubleshooting topics for particular applications, call flow, and network connectivity aspects of Cisco IPC Express systems.

Chapter 17. Troubleshooting Advanced Cisco CME Features

This chapter covers the following topics:

- Debugging and correcting dial plan patterns
- Fixing call transfer patterns
- Troubleshooting call conferencing
- Correcting music on hold (MOH) problems
- Troubleshooting directory services
- Working with the Class of Restriction (COR) feature

Chapter 16, "Troubleshooting Basic Cisco IPC Express Features," explained the basic operation of the Cisco CallManager Express (CME) system. It also looked at some of the problems you might encounter during the registration of IP phones, configuration of network technologies such as Dynamic Host Configuration Protocol (DHCP) and virtual LANs (VLANs), and basic considerations for the correct operation of the Cisco IPC Express GUI. At this point, your phones should be successfully registered to the Cisco CME system and operational.

The next step is to expand your basic two-phone system to a completely functional telephony system suited to your needs. The requirements can vary from system to system, so the potential problems also vary. This chapter explores the issues that may arise while adding configuration and features to a basic Cisco CME system.

Before learning about advanced Cisco CME troubleshooting, one of the most important concepts to understand is the difference between an ephone and an ephone-dn and how these fit into the Cisco IOS software voice architecture. Chapter 5, "Cisco CME Call Processing Features," covered the basic building blocks of a Cisco CME system; you should read it before this chapter. In review

- An ephone (Ethernet phone) is the configuration of the physical device.
- An ephone-dn is the configuration of the phone line or extension. Each ephone-dn has a virtual voice port and dial peer as its subcomponents.

The need to troubleshoot Cisco CME is most likely to arise out of features such as the following:

- The dialplan patterns and transfer patterns associated with the system configuration
- Call transfer and conference scenarios

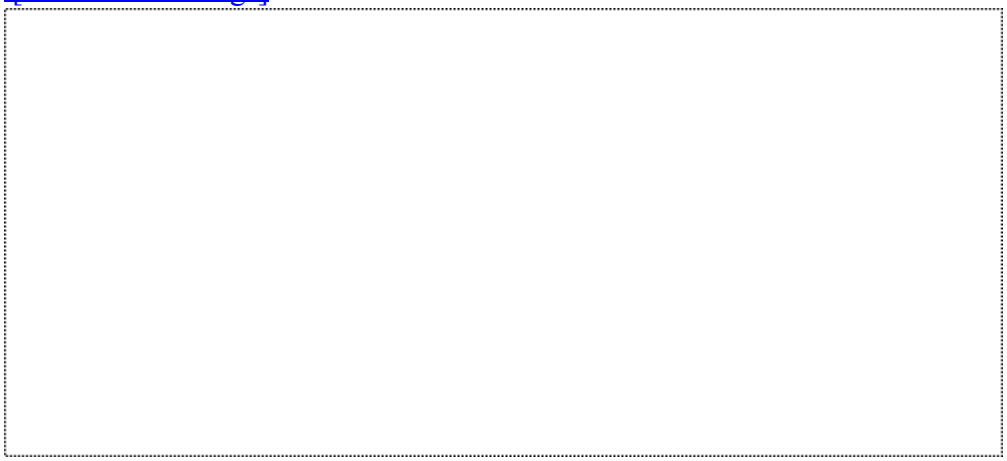
Dialplan Pattern Configuration Problems

A dialplan pattern is an essential element of the Cisco CME configuration to integrate the Cisco CME and PSTN dial plans. If the dialplan patterns are not properly configured, calls to or from the PSTN may not terminate correctly. This section describes some of the common pitfalls in configuring dialplan patterns and how to troubleshoot them.

Figure 17-1 shows the Cisco CME configuration for a company with two geographically separate headquarters and customer support center offices, with Cisco CME deployed at both sites. There are 20 employees in the headquarters office and five employees in the customer support center. Calls between the two sites are routed over the IP WAN using voice over IP (VoIP). This saves the company money and also provides the convenience of uniform four-digit dialing between the sites. Both offices have PSTN connectivity via a Primary Rate Interface (PRI) line.

Figure 17-1. System Configuration

[View full size image]



In Figure 17-1, the extension numbers for the support office start at 1001, with an E.164 prefix of 555111, making the PSTN Direct Inward Dial (DID) number for employees at the office start at 555.111.1001. The E.164 prefix for the headquarters site is 555222. Mapping the DID number to the extension number is achieved by using the dialplan-pattern command.

In this scenario, everything seems to be working fine for the company. Employees can call each other by dialing the appropriate four-digit extension number, and PSTN customers can call the support center by dialing the full PSTN DID number. Example 17-1 shows the relevant portions of the configuration of the headquarters Cisco CME router (HQ_router).

Example 17-1. Headquarters Cisco CME Configuration

```
HQ_Router#show running-config
dial-peer voice 1 voip
 destination-pattern 1...
 session target ipv4:10.10.10.1
 dtmf-relay h245-alphanumeric
!
telephony-service
 load 7960-7940 P00303020214
 max-ephones 24
 max-dn 48
 ip source-address 10.10.10.2 port 2000
 create cnf-files version-stamp 7960 Jan 10 2004 14:22:22
 dialplan-pattern 1 55522220.. extension-length 4 extension-pattern 20..
 max-conferences 4
 transfer-pattern 1...
!
```

Missing Transfer Patterns

Call transfer is one of most commonly used functions in Cisco CME. Transferring calls between multiple Cisco CME systems or to PSTN destinations requires the necessary configuration. This section describes some of the common mistakes made during system configuration (when integrating multiple sites) that prevent call transfer from working.

Call Transfer Doesn't Work

With further business growth, the company adds another office in a new location, and a third Cisco CME system is installed in this location. Figure 17-2 shows the new network configuration for the company with this additional office.

Figure 17-2. System Configuration with the New Office

[View full size image]

The configuration of the Cisco CME in the new location is not much different from the customer support site, except for the numbers assigned to the ephone-dns and the local PSTN DID number. Appropriate dial peers are added to the headquarters and customer support sites for full connectivity. The relevant dial peer configurations for all three Cisco CME routers are shown in the following examples. Example 17-4 shows the dial peer configuration for the headquarters site.

Example 17-4. Headquarters Site Configuration

```
HQ_Router#show running-config
dial-peer voice 1 voip
 destination-pattern 1...
 session target ipv4:10.10.10.1
 dtmf-relay h245-alphanumeric
!
dial-peer voice 2 voip
 destination-pattern 3...
 session target ipv4:10.10.10.2
 dtmf-relay h245-alphanumeric
```

Example 17-5 shows the configuration for the customer support center site.

Example 17-5. Customer Support Center Configuration

Conference Failures

This section examines troubleshooting techniques for the conference feature, which is one of the most frequently used features in any typical office environment. Two common issues related to conference are discussed in this section:

- Conference failure because of unavailable lines
- Conference failure because of a codec mismatch

Conference Failure Because of Unavailable Lines

Example 17-9 shows the configuration of CS_router with the phones for the additional employees added to the system. With this configuration in place, when CS Engineer1 tries to initiate a conference with fellow support engineers, it fails. After making the first call when the conference softkey is pressed, CS Engineer1 gets a "No line available" message on the phone display.

Example 17-9. Configuration of CS_router with Additional Phones

```
CS_router#show running-config
!Output omitted for brevity
...
ephone-dn  1
 number 1001
 name CS Engineer1
 call-forward all 3001
!
ephone-dn  2
 number 1010
 name CS Engineer2
!
ephone-dn  3
 number 1003
 name CS Engineer3
!
ephone  1
 mac-address 0007.EB46.299E
 type 7960
 button  1:1
!
ephone  2
 mac-address 0003.E373.76FB
 type 7960
 button  1:2
!
ephone  3
 mac-address 0030.94C2.9919
 button  1:3
!
ephone  4
 mac-address 000D.BDBE.F372
 button  1:4
```

The debug ephone detail output for the phone of CS Engineer1 is shown in Example 17-10.

Example 17-10. debug ephone detail Output During a Conference Setup Failure

```
CS_router#debug ephone detail
!Output omitted for brevity
```

Unable to Hear Music on Hold

The music on hold (MOH) feature provides a mechanism for the contents of a file stored on the router's Flash to be played to a remote phone when a call is put on hold. If a file does not exist, a short beep (tone on hold) is played to the held party. For IP phones local to the Cisco CME router, the phone itself generates the tone on hold. Hence, even if a music file is present on the router's Flash, a call directly between two IP phones local to the Cisco CME system when put on hold hears only short, intermittent beeps.

Debugging the MOH Problem

The company's system administrator wants to add the MOH feature to both the headquarters and customer support sites. This way, both customers and employees can hear music instead of tones when put on hold. A file, in .au format, is copied to the router's Flash, and the appropriate configurations are added. Example 17-15 shows the relevant Cisco CME configuration.

Example 17-15. show flash Output and MOH Configuration for Cisco CME

```
CS_router#show flash:
-#- --length-- ---------date/time--------- path
1       474338 Dec 5 2002 07:03:14 +00:00 minuet.au
CS_router#show running-configuration
!Output omitted for brevity
...
telephony-service
 load 7960-7940 P00303020214
 max-ephones 24
 max-dn 24
 ip source-address 10.10.10.1 port 2000
 timeouts ringing 30
 url authentication http://10.10.10.1/CCMCIP/authenticate.asp
 create cnf-files version-stamp 7960 Dec 05 2002 03:25:07
 dialplan-pattern 1 555111100. extension-length 4 extension-pattern 100.
 max-conferences 8
 moh minuet.au
```

With the configuration in place, customer calls arriving from the PSTN can hear the MOH. However, calls between the headquarters and customer service sites, when put on hold, still hear a beep tone instead of music.

To analyze this problem, a useful debug command is debug ephone moh. The output of this command on the CS_router is shown in Example 17-16 when a call from headquarters is put on hold.

Example 17-16. Output of **debug ephone moh**

```
CS_router#debug ephone moh
CS_router#
02:34:51: No MOH resources available for DN 1 chan 1 codec 11
CS_router#
CS_router#
02:35:38: Added DN 1 chan 1 to MOH in position 0 with 1 clients
02:35:38: Startup MOH/media streams MOH:1 Media:0
02:36:08: Removed DN 1 from MOH 0 clients at position 0
02:36:08: Shutdown MOH/media streams, no active clients
```

The debug output segment in Example 17-16 shows the output of the debug ephone moh command for a PSTN call put on hold and then retrieved from hold again after some time. From the debug, you can see the ephone-dn listed as one of the MOH stream recipients.

Missing Directory Services Option

One of the features on a Cisco CME system is a real-time directory search function on the phone. The search pattern can be directly entered on the phone keypad, and matching results are displayed. Directory services can be accessed by pressing the directories button on the phone and selecting the appropriate options.

Consider, for example, that when the Directories button is pressed on one of the phones connected to the CS_router, you see the options for missed calls, received calls, and placed calls, but no options for directory services. This section describes how to pinpoint and fix this problem.

Chapter 16 covers downloading a file named SEPxxxxyyyyzzzz.cnf.xml to the phone. This file contains information needed for phone registration. The URL for directory access is one of the pieces of information contained in this file. This file is stored on the router system memory and can be viewed by issuing the command show telephony-services tftp-bindings. The output of this command for the CS_router is shown in Example 17-17.

Example 17-17. Output of **show telephony-service tftp-bindings**

```
CS_router#show telephony-service tftp-bindings
tftp-server system:/its/SEPDEFAULT.cnf
tftp-server system:/its/SEPDEFAULT.cnf alias SEPDefault.cnf
tftp-server system:/its/XMLDefault.cnf.xml alias XMLDefault.cnf.xml
tftp-server system:/its/ATADefault.cnf.xml
tftp-server system:/its/united_states/7960-tones.xml
 alias United_States/7960-tones.xml
tftp-server system:/its/united_states/7960-font.xml
alias English_United_States/7960-font.xml
tftp-server system:/its/united_states/7960-dictionary.xml alias
  English_United_States/7960-dictionary.xml
tftp-server system:/its/united_states/7960-kate.xml
alias English_United_States/7960-kate.xml
tftp-server system:/its/united_states/SCCP-dictionary.xml alias
  English_United_States/SCCP-dictionary.xml
tftp-server system:/its/XMLDefault7960.cnf.xml
alias SEP0007EB46299E.cnf.xml
tftp-server system:/its/XMLDefault7960.cnf.xml
alias SEP0003E37376FB.cnf.xml
```

In Example 17-17, you can see that a filename is associated with the MAC address of each phone on the system. The individual files are aliases of the original XMLDefault7960.cnf.xml file, which is the XML file created for the Cisco 7960 IP Phones. You can see this file's contents by issuing the command more followed by the filename with the path. In the example, the more system:/its/XMLDefault7960.cnf.xml command should return the file's contents. The output of this command for the SEPxxxxyyyyzzzz.cnf.xml file is shown in Example 17-18.

Example 17-18. Contents of the XMLDefault7960.cnf.xml File

```
CS_router#more system:/its/XMLDefault7960.cnf.xml
<device>
<devicePool>
<callManagerGroup>
<members>
<member  priority="0">
<callManager>
<ports>
<ethernetPhonePort>2000</ethernetPhonePort>
</ports>
<processNodeName>10.10.10.1</processNodeName>
</callManager>
</member>
</members>
</callManagerGroup>
```

Working with the Class of Restriction Feature

This section examines issues you may face while configuring the Class of Restriction (COR) feature for Cisco CME phones. COR is an existing Cisco IOS feature used to apply restrictions on originating and terminating calls from the router. COR is a feature applied to voice dial peers. Because dial peers are a subcomponent of the Cisco CME ephone-dn configuration, COR configurations can also be applied to ephone-dns.

First this section offers a brief discussion of COR's operating principles. Then you'll learn how to apply COR to a Cisco CME system and troubleshoot some of the possible errors that may come up. COR is like a lock-and-key mechanism. If the COR applied to an incoming dial peer for incoming calls is a superset of, or equal to, the COR applied to the outgoing dial peer for outgoing calls, the call is allowed to complete successfully.

Figure 17-3 shows a simple setup with two phones illustrating the relationship between the ephone-dn, dial peer, and voice port. Consider a call from Phone A to Phone B. From the perspective of the voice port associated with ephone-dn 1 attached to Phone A, this is an incoming call. Similarly, it is an outgoing call from the perspective of the voice port associated with ephone-dn 2 attached to Phone B.

Figure 17-3. Call Setup from a Voice Port Perspective

Even though the call is an incoming call for Phone B, it is going out of the voice port associated with ephone-dn 2. This concept is very important to understanding COR configurations. The dial peer associated with ephone-dn 1 is the incoming dial peer, and the dial peer associated with ephone-dn 2 is the outgoing dial peer. The key point here is the voice port: the configuration is done from a voice port perspective, not from a phone perspective.

The next thing to clarify is how COR configurations are applied to an incoming or outgoing call. Assume you need a configuration to restrict calls between Phone A and Phone B. To achieve this, two configurations are required:

- One to restrict calls from Phone A to Phone B
- One to restrict calls from Phone B to Phone A

Restrictions always apply in one way only. Hence, COR configurations are needed individually for both incoming and

Summary

In this chapter you learned how to troubleshoot some of the more sophisticated features commonly used in Cisco CME, such as dialplan patterns, call transfer and conference, MOH, directory services, and COR. Some of the important debugs and show commands, such as debug ephone detail, debug ephone moh, and show ephone, were also discussed in detail. The output of these debug commands is a very useful tool for troubleshooting other Cisco CME features as well.

The next chapter focuses more on issues related to the integration of Cisco CME with Cisco Unity and H.323 gatekeepers.

Chapter 18. Troubleshooting Cisco CME Network Integration

This chapter covers the following topics:

- Integrating Cisco CallManager Express (CME) with Cisco Unity voice mail
- Using a centralized Cisco Unity system with a network of Cisco CMEs
- Troubleshooting call transfers and call forwards
- Troubleshooting transcoding
- Troubleshooting H.323 GK integration

In the preceding chapters, you learned about the Cisco CallManager Express (CME) architecture, as well as how to implement and troubleshoot its different features.

This chapter discusses advanced troubleshooting topics in which Cisco CME is networked with other systems, such as other Cisco CMEs at multiple sites in your network and with a Cisco Unity system for voice mail. This chapter also discusses troubleshooting supplementary services, such as call transfer and call forward and integrating Cisco CME with an H.323 gatekeeper (GK).

Integrating Cisco CME with Cisco Unity Voice Mail

As discussed in Chapter 11, "Cisco CME External Voice Mail Options," Cisco Unity is a unified messaging application running on Windows 2000. It uses the Skinny Client Control Protocol (SCCP) to integrate with Cisco CME. This section discusses troubleshooting the integration of Cisco Unity with Cisco CME.

Configuring Cisco CME for Cisco Unity

Cisco Unity connects to Cisco CME as an SCCP device. You configure ephone-dns and ephones for Cisco Unity ports to register with Cisco CME. Cisco Unity uses its Windows Telephony Application Programming Interface (TAPI) to integrate with Cisco CME. A Telephony Service Provider (TSP) is implemented to interface with Cisco CME and provide call control services to the Cisco Unity application. Hence, the Cisco Unity TSP and Cisco CME configuration should match.

Example 18-1 shows a sample Cisco CME configuration for a five-port Cisco Unity system.

Tip

This configuration is used in the next troubleshooting sections, so you might want to bookmark this page until you finish this section on Cisco Unity integration.

Example 18-1. CME Configuration for Cisco Unity to Integrate Using SCCP

```
cme#show running-config
ephone-dn 8
 number 9050
 preference 0
 huntstop
 mwi on
!
ephone-dn 9
 number 9051
 preference 0 secondary 9 huntstop
 huntstop
 mwi off
!
ephone-dn 10 dual-line
! We need dual-line for Unity to execute consult transfers at its Auto Attendant number
8050
 name Voicemail 1
 preference 1 secondary 9
 no huntstop
!
ephone-dn 11 dual-line
 number 8050
 name Voicemail 2
 preference 2 secondary 9
 no huntstop
!
ephone-dn 12 dual-line
 number 8050
 name Voicemail 3
 preference 3 secondary 9
 no huntstop
!
ephone-dn 13 dual-line
 number 8050
 preference 4 secondary 9
 huntstop
!
ephone-dn 14
```

A Cisco Unity System with a Network of Cisco CMEsCentralized Voice Mail Architecture

Cisco Unity can support thousands of subscribers, so when it is used with Cisco CME, it often makes sense to share a single Cisco Unity system among many Cisco CMEs in a network of Cisco CME nodes. The advantages of doing this instead of using a Cisco Unity Express (UE) system at each site are that voice mail administration can be done at a single central location, and a voice mail networking configuration is not needed to exchange messages between users located at different sites. However, you still need a Cisco CME collocated with Cisco Unity because Cisco Unity registers with only a single local Cisco CME system. This configuration was discussed in Chapter 11.

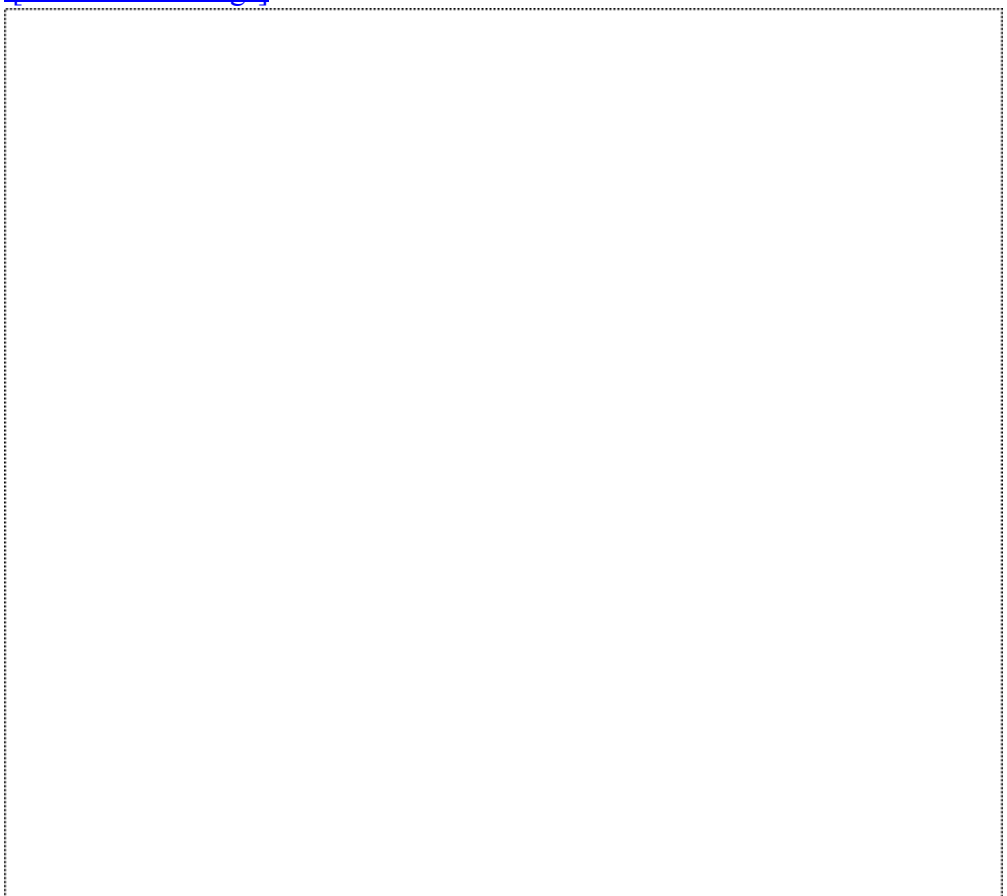
A centralized Cisco Unity system also means that all calls that access voice mail have to travel over the WAN. This requires a quality of service (QoS)-enabled WAN backbone as well as more bandwidth, especially if Cisco Unity is configured to use the G.711 codec. You can reduce bandwidth consumption in several ways:

- Configure Cisco Unity to use the G.729 codec. In this case, Cisco Unity can accept both G.729 and G.711 calls.
- Use a hardware transcoder in front of Cisco Unity to convert G.729 audio streams to G.711. Cisco CME 3.2 and greater supports a transcoding function.

One more thing to consider in centralized voice mail architecture is that all the IP phone extensions in the network must be unique. In other words, no two Cisco CMEs can share the same phone extension. Also, if you use central Cisco Unity as the AA for all the Cisco CME sites, you need to enable H.450 services in the network for call transfers to extensions appearing on different Cisco CMEs, and configure transfer patterns on the central Cisco CME. Figure 18-5 shows a sample topology of such a network.

Figure 18-5. Centralized Cisco Unity with Distributed Cisco CME Sites

[View full size image]



Troubleshooting Call Transfers and Call Forwards

Chapter 7, "Connecting Multiple Cisco CMEs with VoIP," covered the call transfer and forward mechanisms and protocols used to implement network call transfers using H.323 and SIP. It is recommended that you review Chapter 7 to effectively troubleshoot any problems in this area. This section discusses how to troubleshoot issues with local and network call transfers and forwards and describes how to detect and fix common configuration issues.

Troubleshooting Call TransfersTransfer Attempts Get Reorder Tone

Transfers to local extensions on a Cisco CME are enabled by default. For all the other transfers, including those to local PSTN numbers, you must configure transfer patterns. Sometimes transfer attempts may not work because either a transfer pattern is not configured or the transfer target's number may not match any transfer patterns configured on the system.

The easiest way to fix this situation is to use the transfer-pattern .T command, which allows transfers to all numbers. However, this solution might be undesirable if you are concerned about fraudulent use of call transfers. To see if a transfer pattern is matched, turn on the Cisco CME debug ephone state command to see which ones are not matching, and fix them with modifications to the transfer patterns for the desired transfer targets. This is shown in the debugs in Example 18-13. The user presses the transfer softkey when the call is in the connected state, gets dial tone, and starts dialing the transfer-to number. However, no transfer pattern matches this nonlocal number, so the caller hears reorder tone.

Example 18-13. Missing Transfer Pattern

```
CME#debug ephone state
SkinnyTryCall to 8010 instance 1 match DN 1
Feb 22 18:29:42: ephone-1[2]:Call Info DN 1 line 1 ref 244 called 8010 calling
   4085555002 origcalled 8010 calltype 1
Feb 22 18:29:42: ephone-1[2]:Call Info for chan 1
Feb 22 18:29:42: ephone-1[2]:Original Called Name Waugh
Feb 22 18:29:42: ephone-1[2]: lleyton hewitt calling
Feb 22 18:29:42: ephone-1[2]: Waugh
Feb 22 18:29:47: ephone-1[2][SEP003094C29D3C]:SoftKeyEventMessage event 4
  line 1 callref 244
Feb 22 18:29:47: ephone-1[2]:SK TRANSFER line 1 ref 244
Feb 22 18:29:47: ephone-1[2]:TransferButtonPress
Feb 22 18:29:47: ephone-1 TRANSFER using transfer-system full DN 1
  consult transfer
Feb 22 18:29:47: SkinnyGetCallState for DN 1 chan 1 CONNECTED
Feb 22 18:29:47: called DN -1 chan 1, calling DN -1 chan 1 phone 1
  incoming s2s:0
Feb 22 18:29:47: SkinnyGetCallState for DN 1 chan 1 CONNECTED
Feb 22 18:29:47: called DN -1 chan 1, calling DN -1 chan 1 phone 1
  incoming s2s:0
Feb 22 18:29:47: SkinnyGetCallState for DN 3 chan 1 IDLE
Feb 22 18:29:47: called DN 2 chan 1, calling DN -1 chan 1 phone 1 s2s:0
Feb 22 18:29:47: ephone-1[2]:Transfer with consult line 1 DN 1 chan 1
  using second line 2 DN 1 chan 1
Feb 22 18:29:47: ephone-1[2]:Consult for line 1 DN 1 chan 1 using
  line 2 DN 3 chan 1
Feb 22 18:29:47: ephone-1[2]:HoldButtonPress (allow_toggle = 0)
Feb 22 18:29:47: SkinnyGetCallState for DN 1 chan 1 CONNECTED
Feb 22 18:29:47: called DN -1 chan 1, calling DN -1 chan 1 phone 1
  incoming s2s:0
Feb 22 18:29:47: ephone-1[2]:HoldButtonPress HOLD activated for DN 1 chan 1
Feb 22 18:29:47: ephone-1[2]:HoldButtonPress DN 1 chan 1 other DN -1 chan 1
   other ephone-(-1)
Feb 22 18:29:47: ephone-1[2]:Binding ephone-1 to DN 1 chan 1 s2s:0
Feb 22 18:29:47: Adding DN 1 chan 1 to MOH
.........
Feb 22 18:29:47: ephone-1[2]:CloseReceive
```

Troubleshooting Transcoding

Cisco CME 3.2 introduces support for transcoding services. You need transcoding services when you have devices and applications in the network that don't support lower-bandwidth codecs, and you don't have enough bandwidth available in your network. In such a case, the devices that don't support lower-bandwidth codecs are colocated with a transcoding device that converts media streams of different codecs. This section discusses the configuration and troubleshooting of such transcoding devices.

Configuring Transcoding Services

One key example of a transcoding scenario is when Cisco UE terminates AA or voice mail calls that must be G.711. If these calls originate from other sites across your WAN, they likely use G.729 to conserve bandwidth. For such calls to divert successfully into Cisco UE, transcoding services are needed. Also, if Cisco CME has to establish a three-party conference in which one of the endpoints is on a G.729 gateway, a transcoder is needed to complete and mix the conference. This section discusses the configuration of a transcoding device for Cisco CME and how to troubleshoot issues relating to transcoding services.

Assume in the topology given earlier in Figure 18-6 that Cisco CME 2 has a high-density voice network module (NM-HDV2) with digital signal processors (DSPs) that can be configured for transcoding services. Example 18-29 shows such a configuration with the NM-HDV2 present in slot 4 of the Cisco CME 2 router.

Example 18-29. DSP Farm Configuration for Transcoding Services

```
CME2#show running-config
voice-card 4
 dspfarm
 dsp services dspfarm
```

The DSP farm (for transcoding services) has to register with Cisco CME. The DSP farm does not need to be physically present in the Cisco CME router chassis with which it is registered. It may be present in a separate PSTN gateway router chassis at the same site. In this example, however, the DSP farm is hosted on the Cisco CME router itself, which is a common deployment.

Example 18-30 shows the configuration required to register the DSP farm to Cisco CME 2. You have to identify an interface whose IP address can be used in media setup for the services. In Example 18-30, FastEthernet interface 0/0 is used. The Cisco CME IP address that this transcoding device registers with is also configured (192.168.0.1 in Example 18-30). A transcoding profile is created in which you mention all the codecs that should be transcoded. Several commands start with sccp, because the transcoding device uses SCCP to register with Cisco CME, and it is controlled by using SCCP.

Example 18-30. Configuration to Register the DSP Farm to a Cisco CME

```
CME2#show running-config
sccp local FastEthernet0/0
sccp ccm 192.168.0.1 identifier 1
sccp ip precedence 3
sccp
!
sccp ccm group 1
 bind interface FastEthernet0/0
 associate ccm 1 priority 1
 associate profile 1 register mydsp1
!
dspfarm profile 1 transcode
 codec g711ulaw
 codec g711alaw
 codec g729ar8
```

Troubleshooting H.323 GK Integration

The section "Troubleshooting Common Problems with Network Call Transfers" covered common problems with call routing (dial peer) configurations. You can manage routing configuration between nodes in a network effectively if you have only a handful of Cisco CME sites in your network. As the number of Cisco CMEs and PSTN gateways increases, it becomes increasingly cumbersome to use manual dial peers pointing to each Cisco CME and PSTN gateway in the network at each of the sites. This is where a GK becomes a useful addition to the network. You can configure a Cisco IOS voice-enabled router as an H.323 GK by using the appropriate Cisco IOS image.

Figure 18-7 shows a sample topology with four Cisco CMEs in the network and using one of the Cisco CMEs as a GK. With this kind of network, all the call routing information can be centralized in the GK. All the extension-based and E.164 dial peers on the remote Cisco CME systems point to the GK. Only the GK has true knowledge of the dialing plan. This minimizes issues with call transfers and forwards caused by a lack of appropriate call routing dial peers on all Cisco CMEs.

Figure 18-7. Topology for the GK in a Cisco CME Network

[View full size image]

Example 18-36 shows the relevant part of the GK configuration. The name of the zone containing all the Cisco CMEs is gk.

Example 18-36. GK Configuration

```
CME4#show running-config
gatekeeper
 zone local gk cisco.com 192.168.0.1
 no shutdown
```

All the PSTN gateways and remote Cisco CME systems must be configured as shown in Example 18-37. Usually the

Summary

This chapter covered advanced troubleshooting topics, including Cisco CME integration with Cisco Unity for voice mail and AA and various call transfer scenarios across multiple Cisco CME nodes in an H.323 network. H.450 operation and troubleshooting were discussed, as well as scenarios that require transcoding of codecs to complete a call flow. This chapter also covered network topologies with GK configurations.

Chapter 19. Troubleshooting Cisco UE System Features

This chapter covers the following topics:

- General Cisco Unity Express (UE) system troubleshooting techniques
- Troubleshooting problems that might occur during system installation
- Troubleshooting Cisco UE startup problems
- Troubleshooting the Cisco UE Initialization Wizard
- Troubleshooting the system backup and restore operations

The previous chapters in Part IV, "Maintenance and Troubleshooting," took a closer look at Cisco CallManager Express (CME) troubleshooting of the call processing and phone components of your system, as well as some voice mail scenarios with external voice mail systems such as Cisco Unity. This chapter takes an in-depth view of troubleshooting Cisco CME's integrated voice mail application, Cisco Unity Express (UE).

Cisco UE's hardware and software architecture was discussed in Chapter 3, "Cisco IPC Express Architecture Overview." This fundamental architectural knowledge is required to understand the troubleshooting tools and techniques discussed in this chapter. In Chapter 3, you learned how a Cisco IOS router interfaces with the Cisco UE hardware module using the Router Blade Communication Protocol (RBCP), which is the fundamental mechanism used as the communication path between Cisco UE and the Cisco CME router and is integral to understanding later sections in this chapter.

This chapter covers troubleshooting the system-level aspects of Cisco UE, such as general problem isolation techniques, installation, startup, and backup and restore.

General Troubleshooting Techniques

Cisco UE offers several types of tools to help you isolate problems on the system:

- Viewing system information and status (using a variety of show commands)
- Event logging to log files including information, warning, error, and fatal messages
- A tracing facility showing detailed information on system events, information, and decisions (similar to Cisco IOS debug information)

Viewing Generic System Information

This book provides you with information about Cisco UE so that you can troubleshoot and fix most of the common problems you might encounter. In case you still have to contact the Cisco Technical Assistance Center (TAC) for additional support, however, it is helpful to provide the Cisco TAC with some basic information about your system. This basic information includes the software version, the software license, the current configuration of the Cisco CME and Cisco UE components, and some information about the current usage of the system. This section describes how to view that basic information.

Example 19-1 uses the show version command to provide information about the hardware form factor of the Cisco UE module, the model type of the router that is hosting the Cisco UE, the speed with which CPU is running, and additional hardware-specific information. One thing to note is that the AIM-CUE runs at 150 MHz if it is hosted on 2600XM or 2691 series access routers. In other routers, it runs at 300 MHz.

Example 19-1. Displaying the System Version

```
CUE#show version
CPU Model:              Celeron (Coppermine)
CPU Speed (MHz):        154.926
CPU Cache (KByte):      128
Chassis Type:           C2600
Chassis Serial:         FSJC0400673
Module Type:            AIM
Module Serial:          FHH07330001
```

Example 19-2 uses the show software version command to display the versions of the Cisco UE software components installed on the system. Identifying the installed software versions is important when troubleshooting interworking issues. For example, some Cisco UE features might be supported only with a certain version of Cisco CME or Cisco UE.

Example 19-2. Displaying the Software Version

```
CUE#show software version
Installed Packages:
 - Bootloader (Primary) 1.0.17
 - Global 2.0.0.12
 - Voice Mail 2.0.0.12
 - Bootloader (Secondary) 2.0.0
 - Core 2.0.0.12
 - Auto Attendant 2.0.0.12

Installed Languages:
 - US English 2.0.0.0
```

Troubleshooting Installation Problems

When you install a Cisco UE module, problems might occur in downloading the software package. These problems might be caused by network connectivity or package issues. This section discusses some common problems that might occur during software installation of Cisco UE and ways to troubleshoot them.

Network Connectivity Issues

If the Cisco UE module is unable to establish contact with the FTP server where the software load resides, the error shown in Example 19-13 occurs when you attempt to install the software.

Example 19-13. Installation Failed

```
cueinstaller#>software install package url ftp://username:password@1.3.61.61/
  cue-vm.1.1.0.6.pkg
RAMDisk mounted
Connecting to host...
curl: (7) Connect failed
ERROR: Host did not respond.
Please check the host ip and try again.
RAMDisk unmounted
```

First ensure that the IP address of the FTP server is correct. Verify all the parameters given in the install command. If you are sure these are all correct, verify the IP connectivity from the Cisco UE module to the router. Reboot the Cisco UE module, as shown in Example 19-14, and press *** at the first prompt. This action takes you to the bootloader prompt.

Example 19-14. Checking IP Connectivity Between Cisco UE and the FTP Server

```
cueinstaller#>reboot
WARNING: This will reboot the Service Engine!
Do you wish to continue (y,n) [n] y
```

The bootloader has a ping command, as shown in Example 19-15.

Example 19-15. Checking IP Connectivity from the Bootloader

```
ServicesEngine boot-loader> ping 1.3.61.61
Sending 5, 32 byte ICMP Echos to 1.3.61.61:
.....
Success rate is 0% (0/5)

ServicesEngine boot-loader> ping 1.3.61.16
Sending 5, 32 byte ICMP Echos to 1.3.61.16:
!!!!!
Success rate is 100% (5/5)
```

If the Cisco UE system cannot ping the FTP server, you might have the wrong configuration of IP parameters in the bootloader. Example 19-16 shows how to check the bootloader configuration. If you see anything wrong, you can use the bootloader config command to make modifications.

Example 19-16. Checking the Bootloader Configuration

```
ServicesEngine boot-loader>show config
IP addr:                1.4.13.90
Netmask:                255.255.0.0
```

Troubleshooting Cisco UE Startup

You might experience situations in which the basic IP communication between the host router and the Cisco UE module fails even after proper configuration of the host router. You can identify such a problem by looking for waiting events such as those shown in an extract of the installation output in Example 19-20.

Example 19-20. Installation Output Extract Showing Cisco UE Waiting for the Host Router

```
==> only eth0 exists, we must be running on an AIM
==> only eth0 exists, we must be running on an AIM

Router communications servers initializing...complete.
Waiting for IOS to register IP address.
- waited 10 seconds...
Waiting for IOS to register IP address.
  - waited 20 seconds...
```

Cisco UE is waiting for commands from the Cisco IOS router to configure Cisco UE with its IP address and default gateway parameters so that Cisco UE can communicate with the rest of the network. However, it is not receiving any response from the router. This process of configuring the Cisco UE module through the host Cisco IOS router uses the RBCP, as discussed in Chapter 3. There might be some situations in which you have to troubleshoot this protocol exchange between the host router and Cisco UE.

When Cisco UE successfully communicates with the router using RBCP and receives its IP parameters, you see the message shown in Example 19-21 on the Cisco UE console during application bootup.

Example 19-21. Correct Response from the Host Router to Cisco UE

```
==> only eth0 exists, we must be running on an AIM
==> only eth0 exists, we must be running on an AIM
Router communications servers initializing...complete.
IOS IP Address Registration complete.
```

The following few sections discuss troubleshooting problems during application bootup and communication issues between Cisco UE and the host router.

Checking Console Output on the Cisco UE Hardware

If you are having a problem opening a session to the Cisco UE module, or you don't see any output on the console, you can use the following command to check the console messages on the Cisco UE module without needing to open a session to the Cisco UE module:

```
router#test service-module service-engine slot/unit console
```

By default, this command displays the most recent 80 lines stored in the console buffer. However, it is possible to specify an offset of greater or less than 80, or to view all the messages stored in the console buffer.

Checking the RBCP Communication

To check the RBCP status on the Cisco UE module from the router, you can use the following test command:

```
router#test scp ping slot
```

This command sends a ping to the Cisco UE module as an RBCP message using operational code (opcode) 0x11. If

Troubleshooting the Initialization Wizard

The Cisco UE Initialization Wizard is a GUI tool provided to complete the installation of the system, as covered in Chapter 13. After new software is loaded on the system, you must run the Initialization Wizard. You invoke a browser on a PC with a URL pointing to Cisco UE. After logging in, you go through a number of steps, including

- Importing users from Cisco CME
- Defining the various pilot numbers

Here are a couple common issues encountered when running the Initialization Wizard:

- You are unable to log on to the Initialization Wizard
- After all the procedures are complete, the system returns with a failure

The next sections describe how to identify both issues.

Unable to Log into the Initialization Wizard

At the login window, you enter the administrator username and password assigned during the post-installation procedures. If you are unable to log in, the issue could be that the username and password were not entered properly or that uppercase or special characters have been used. To verify that the administrator username is legitimate, go to the CLI and check the Cisco UE configuration. The username entered should be one of the administrators defined on the system. You can verify that the username admin belongs to the Administrators group, as shown in Example 19-27.

Example 19-27. Displaying Members of the Administrators Group

```
cue#show group detail groupname Administrators
Full Name:           Administrators
Description:
Phone:
Phone(E.164):
Language:            systemDefault(ga_IE)
Owners:
Members:         admin
Privileges:          superuser ManagePrompts ManagePublicList ViewPrivateList
```

If you have verified that the username is an administrator but you still cannot log in, you can bypass this by assigning a new password for the login, as shown in Example 19-28.

Example 19-28. Defining an Administrator

```
CUE#username temp password temp
CUE#config terminal
CUE(config)#group Administrators member temp
```

temp is any name you choose for this purpose.

Note

Troubleshooting Backup and Restore

This section discusses problem scenarios you might encounter when performing a backup or restore operation of the Cisco UE configuration or voice message data, including the following:

- Cannot find or connect to the backup/restore server.
- Backup or restore cannot start.
- Backup or restore is incomplete.
- The restore operation cannot restore the correct configuration or data.

The next sections describe techniques for troubleshooting each of these issues.

Cannot Find or Connect to the Backup/Restore Server

To start a backup or restore operation, you must first configure the FTP server URL and corresponding directory. Backup and restore are done to the same server location. The CLI to configure the backup server is

```
cue(config)#backup server url ftp://server_ip_address/directory [username
 name password password]
```

Use the following processes to troubleshoot server connectivity problems:

- Check the Cisco UE configuration to ensure that all the backup and restore configuration parameters are correct. The critical parameters are the server's IP address and the appropriate directory path.
- Use the commands show backup server and show backup history to ensure that the configuration is set up properly.
- Ping the FTP server to ensure that it can be reached.
- The backup path specified in the Cisco UE configuration shows the default FTP path. Check the path set on the server to ensure that the proper directory has been selected.

Backup or Restore Cannot Start

The FTP server for backup and restore might require a user login and password (this is an optional but recommended configuration). Also, some FTP servers limit the transferable block of data to a certain size. If a login is configured on your FTP server, verify that you have permission to read and write on the FTP server.

If the backup server has been configured but cannot be reached when the backup process starts, the system shows an error message, as shown in Example 19-32.

Example 19-32. Backup Restore Error Message

Summary

This chapter provided an overview of the troubleshooting tools and techniques to be used with the Cisco UE system. You learned about potential problems with basic connectivity between the host router and the Cisco UE application, including tracing the RBCP messages. Isolating installation, startup, and other connectivity problems were covered next. This chapter also discussed backup and restore functions to show how to troubleshoot problems in this area.

The next chapters cover Cisco UE troubleshooting specific to the AA and voice mail applications.

Chapter 20. Troubleshooting Cisco UE Automated Attendant

This chapter covers the following topics:

- Troubleshooting problems that may occur with a customized automated attendant (AA)
- Tracing a call flow through AA from call arrival at the AA pilot number to the call's transfer to the destination extension

This chapter discusses how to troubleshoot operational and runtime issues with developing and deploying the Cisco Unity Express (UE) customized automated attendant (AA) application.

Chapter 3, "Cisco IPC Express Architecture Overview," provides an overview of the Cisco UE AA and voice mail system components and architecture. As such, it is a good foundation for the topics discussed in this chapter. A key software component of Cisco UE is the Customer Response Solutions (CRS) software. CRS is the infrastructure leveraged by all other Cisco UE applications, including the AA. CRS is also the component used to implement the customized AA applications.

Troubleshooting a Customized AA

As discussed in Chapter 9, "Cisco IPC Express Automated Attendant Options," Cisco UE allows you to create up to five customized AA applications or top-level scripts for each Cisco UE system. Creating and deploying such applications involves the following steps:

Step 1. Establish clear requirements for your AA application, and draw a flow chart for the desired menu flow.

Step 2. Determine the wording of the prompts for the application.

Step 3. Record the prompts using the tool of your choice, such as a PC-based recorder or Cisco UE Administration Via Telephony (AVT) (called the Greeting Management System [GMS] before Cisco UE release 2.1). You also can outsource the creation of the prompts to a recording studio.

Step 4. Name the prompts (.wav files) appropriately.

Step 5. Convert the application flow chart you created in Step 1 into one or more scripts using the Cisco UE AA Script Editor. (Use appropriate variables, and make sure you use the right prompt names in the script.)

Step 6. Validate the scripts by selecting Tools > Validate in the Cisco UE AA Script Editor.

Step 7. Upload the scripts and prompts to the Cisco UE system using the GUI or (CLI).

Step 8. Create an AA application from the Voice Mail > Auto Attendant GUI menu. This allows you to associate the script with a dial-in phone number (the AA pilot number for this script) and prepares the application to take live calls.

Step 9. Ensure that the Cisco CME router has a Session Initiation Protocol (SIP) dial peer with the correct dual-tone multifrequency (DTMF) relay and codec options pointing to the new AA pilot number (called a trigger internally in the Cisco UE software) created in the preceding step.

Even though you have followed all the necessary precautions in creating the AA, some issues may still come up. When issues occur, you should use the troubleshooting techniques discussed in this chapter.

The following sections discuss possible problems you might encounter during the configuration of an AA application. These are not runtime issues. Instead, they are problems you may experience during the configuration or deployment of the system or a new AA menu.

Ensuring a Correct Configuration

An AA application may have multiple physical script files. In this case, a main .aef file calls other .aef files using the Call Subflow step. It is possible that you uploaded the main script but forgot to upload all the subflows needed by this script. Or you might have an undefined variable in the script if you forgot to validate the script in the AA script Editor before uploading.

You can find and fix many such issues when you configure the system instead of finding them during runtime when active user calls are placed to the application. You can find and eliminate most of these issues when creating the AA application using the Cisco UE administration GUI, as shown in Figure 20-1.

Figure 20-1. Creating an AA Application

[View full size image]

Tracing a Call Flow in the AA

The previous sections discussed some problems that may occur when deploying customized AA applications. However, similar issues might be encountered with other applications on Cisco UE because of misconfiguration or interworking difficulties with other applications in the network.

To effectively troubleshoot such unexpected issues, it is helpful to understand the complete operation of the AA from the point when a call arrives at the Cisco UE software to the point where the AA script starts and CRS steps are executed. The sample script that is used to demonstrate this section is available in Appendix B, "Sample Cisco UE AA Scripts." You can also download this script from the Cisco Press website at http://www.ciscopress.com/title/158705180X.

SIP Call Delivered to the AA Pilot Number

Figure 20-2 showed a SIP call flow for a call into Cisco UE. This section describes an incoming SIP call to a particular trigger (the AA pilot number) and the call being answered by the application. You also see how the media is set up using the RTP protocol and how the AA script steps are executed. The example used is a custom AA that does the following:

- Plays a menu to the caller for dial-by-extension and dial-by-name
- Transfers the call to the selected user
- Plays the spoken name if the user selected has recorded one

Example 20-13 shows the trace commands used to troubleshoot issues with SIP calls arriving at the AA. A SIP INVITE is the first message to arrive at the AA trigger.

Example 20-13. Incoming INVITE to the AA

```
cue#trace ccn stacksip dbug
cue#trace ccn managerappl dbug
cue#trace ccn libmedia dbug
cue#trace ccn subsystemcmt dbug
cue#show trace buffer tail
3544 01/12 11:09:14.523 ACCN SIPL 0 -------
INVITE sip:5600@1.3.6.127:5060 SIP/2.0
Via: SIP/2.0/UDP 1.3.6.27:5060;branch=z9hG4bKC96
From: <sip:6019@1.3.6.27>;tag=ADDECF6C-134D
To: <sip:5600@1.3.6.127>
Date: Mon, 12 Jan 2004 19:09:14 GMT
Call-ID: A56DFF23-446911D8-8934AECA-10549FA0@1.3.6.27
Supported: 100rel,timer
Min-SE: 1800
Cisco-Guid: 2755249499-1147736536-2301734602-273981344
User-Agent: Cisco-SIPGateway/IOS-12.x
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, COMET, REFER, SUBSCRIBE,
  NOTIFY, INFO, UPDATE, REGISTER
CSeq: 101 INVITE
Max-Forwards: 6
Remote-Party-ID: <sip:6019@1.3.6.27>;party=calling;screen=no;privacy=off
Timestamp: 1073934554
Contact: <sip:6019@1.3.6.27:5060>
Call-Info: <sip:1.3.6.27:5060>;method="NOTIFY;Event=telephone-event;Duration=2000"
Expires: 180
Allow-Events: telephone-event
```

Summary

In this chapter, you learned how to troubleshoot call execution by the AA scripts in your system. This chapter covered the call flows for the calls coming into the AA and how to troubleshoot common problems at runtime. This chapter also presented the traces of a typical AA call to help you understand how the steps execute and what information to look for in the various traces. At this point, you should be able to troubleshoot most of the problems you might encounter when deploying a Cisco UE AA.

Chapter 21. Troubleshooting Cisco UE Integrated Voice Mail Features

This chapter covers the following topics:

- Common voice mail show commands
- Troubleshooting mailbox graphical user interface (GUI) configuration problems
- Troubleshooting Cisco Unity Express (UE) and Cisco CallManager Express (CME) integration
- Troubleshooting the Telephony User Interface (TUI) and VXML browser
- Troubleshooting the database, Lightweight Directory Access Protocol (LDAP), and mailbox activities
- Troubleshooting the message waiting indicator (MWI)
- Troubleshooting voice mail Voice Profile for Internet Mail (VPIM) networking

This chapter discusses the debugging and tracing aspects of the Cisco Unity Express (UE) voice mail application for Cisco IP Communications (IPC) Express. The discussion starts by looking at configuration problems that show up in the graphical user interface (GUI) and how to address them. Another area of possible problems is the interface between Cisco UE and Cisco CallManager Express (CME), which you also will learn how to troubleshoot. The Voice Browser step of the Cisco Customer Response Software (CRS) is used to implement the Telephony User Interface (TUI), and in this chapter you learn how to interpret voice browser traces and TUI sessions.

Voice mail interacts with back-end data stores such as the Lightweight Directory Access Protocol (LDAP) directory and the Structured Query Language (SQL) database to retrieve user and mailbox information. This chapter covers the troubleshooting of issues that might require you to look at the LDAP and SQL aspects of the voice mail application.

A common problem with voice mail systems includes incorrect operation of the message waiting indicator (MWI), so you also learn about this area of problem isolation. The last area of voice mail troubleshooting covered in this chapter is when multiple Cisco UE systems are networked via Voice Profile for Internet Mail (VPIM).

Chapter 3, "Cisco IPC Express Architecture Overview," gave an overview of the voice mail system components and architecture and provide a good foundation for the topics discussed in this chapter.

Common Voice Mail show Commands

Although tracing is one of the key troubleshooting tools in Cisco UE, it is not always necessary. Many of the show commands are sufficient to identify certain problems. The following sections discuss the most common show commands you can use to pinpoint trouble spots.

Listing Mailboxes with Usage Statistics

The show voicemail mailboxes command gives a very helpful summary of each mailbox. Although you can display the same data for each mailbox in the GUI, no GUI screen provides the same comprehensive system summary that is given by the output of this command-line interface (CLI) command. The show voicemail mailboxes command lists mailbox statistics in table format and contains the following fields:

- Total number of messages
- Number of new and saved messages
- Total time used for the mailbox
- Total mailbox size and the percentage of the mailbox used

Example 21-1 shows sample output of the show voicemail mailboxes command. In this output, Jane Smith's mailbox does not contain any messages, but still the message time (MSGTIME) associated with the mailbox is given as 9 seconds. That is because Jane Smith has recorded one or more greetings, and the time for the greetings is counted as part of the total mailbox message time. However, spoken names are not considered part of the mailbox time, because they are stored in LDAP rather than the SQL database.

Example 21-1. Displaying Mailbox Statistics

```
cue#show voicemail mailboxes
OWNER                          MSGS NEW   SAVED MSGTIME MBXSIZE   USED
"johnsmith"                      4    2     2      60       3000     2 %
"janesmith"                      0    0     0      9        3000     1 %
"group1"                         0    0     0      0        3000     0 %
"group2"                         0    0     0      0        3000     0 %
"group3"                         0    0     0      0        3000     0 %
"group4"                         0    0     0      0        3000     0 %
"group5"                         0    0     0      0        3000     0 %
"johndoe"                        0    0     0      0        3000     0 %
"janedoe"                        0    0     0      14       3000     1 %
```

Displaying Mailbox Details

To display the details of a particular mailbox, use the show voicemail detail mailbox userid command. As shown in Example 21-2, it displays a mailbox's characteristics, including the message counts, date of creation, and last-modified time stamp.

Example 21-2. Displaying Single Mailbox Details

```
cue#show voicemail detail mailbox johndoe
Owner:                                  /sw/local/users/johndoe
Type:                                   Personal
Description:                            johndoe mailbox
```

Troubleshooting Mailbox GUI Configuration Problems

You can select to create mailboxes manually when users or groups are added, or you can add mailboxes in bulk fashion during the Cisco UE Initialization Wizard when user definitions are imported from Cisco CME. The following issues may occur in the GUI:

- Mailbox configuration problems
- Orphaned mailboxes
- User and group configuration problems
- General delivery mailbox problems

Mailbox Configuration

To troubleshoot mailbox configuration problems in the GUI, the mailbox trace command trace webInterface mailbox all is useful.

Example 21-6 shows sample output of this command where the user attempts to access a mailbox that does not exist. The administrator creates a new mailbox, deletes another mailbox, and then verifies that the new mailbox has been created successfully by checking the mailbox GUI page. Other mailboxes are also shown on the GUI page. The PERSONAL_00000000000000000000009 text shown in the trace is an internal unique identifier for each mailbox. You can use this identifier to isolate operations and events that pertain to a particular mailbox in the system.

Example 21-6. Mailbox Trace

```
cue#trace webInterface mailbox all
cue#show trace buffer tail

2607 02/28 19:39:30.308 webI mail 1 Attempt to read mailbox
  PERSONAL_00000000000000000000006
2178 02/28 19:40:23.483 webI mail 1 Error while reading Mailbox by owner. no such
 attribute
2178 02/28 19:40:23.563 webI mail 2 Created mailbox:
 PERSONAL_00000000000000000000009
2178 02/28 19:40:23.564 webI mail 2 Proceeding to Populate other values for new
  mailbox
2178 02/28 19:40:23.564 webI mail 2 Saving Mailbox
2178 02/28 19:40:23.625 webI mail 2 Finished populating other values for new
  mailbox
2667 02/28 19:40:23.893 webI mail 1 Filter for mailboxes : (&(ownerDn=*))
2667 02/28 19:40:23.899 webI mail 1 Attempt to read mailbox
  PERSONAL_00000000000000000000004
2667 02/28 19:40:23.902 webI mail 1 Attempt to read mailbox
  GENERAL_00000000000000000000000
2667 02/28 19:40:23.904 webI mail 1 Attempt to read mailbox
  PERSONAL_00000000000000000000008
2667 02/28 19:40:23.906 webI mail 1 Attempt to read mailbox
  PERSONAL_00000000000000000000002
2667 02/28 19:40:23.909 webI mail 1 Attempt to read mailbox
  PERSONAL_00000000000000000000009
2667 02/28 19:40:23.918 webI mail 1 Attempt to read mailbox
  PERSONAL_00000000000000000000005
2667 02/28 19:40:23.919 webI mail 1 Attempt to read mailbox
```

Troubleshooting Cisco CME and Cisco UE Integration

Cisco CME is the system's call control engine. All calls coming into voice mail to leave or retrieve messages involve an interaction (via SIP) between Cisco CME and Cisco UE. Cisco CME integration with Cisco UE voice mail includes the following activities:

- Forwarding of a call to Cisco UE on busy or ring-no-answer
- Dual-tone multifrequency (DTMF) relay to interact with the voice mail system
- A call transfer from the automated attendant (AA) that forwards into voice mail
- Turning the MWI on or off

Call forwarding into voice mail requires passing the right original called number to the voice mail system so that the correct user's mailbox is selected to play the greeting and leave a message. DTMF relay encompasses the conversion of IP phone keypad digit presses sent via Skinny Client Control Protocol (SCCP) messages from the phone to Cisco CME call control, and from there to Cisco UE via SIP NOTIFY messages. Call transfer from the AA is implemented as a blind transfer using the SIP BYE/Also message sequence.

The most common problems encountered with calls into voice mail are caused by incorrect or incomplete configuration:

- The wrong mailbox selection or no mailbox selection on call forward to voice mail (which might also result in the wrong greeting being played to the caller)
- Digit presses from the phone do not get the required response (in other words, DTMF messages do not reach the application)
- Call transfers from the AA do not work
- Fast-busy tone is heard for calls going to voice mail
- MWI does not operate properly

Correcting each of these problems is discussed in the following sections.

Wrong Mailbox Selection or Unexpected Greeting

Calls to a Cisco CME IP phone can arrive from two different sources:

- A call received from an internal extension, where the dialed number is the phone's extension
- A call received from the PSTN or from another site (routed via H.323 or SIP), where the called number may be the phone's E.164 number

Troubleshooting the TUI and VXML Browser

The main interface used to interact with a voice mail system such as Cisco UE is the Telephony User Interface (TUI). When a call arrives at a voice mail system, the system starts playing specific prompts depending on the call type (for example, an external caller, a subscriber, or a call forwarded into voice mail). To interact with the system, the user presses DTMF digits on his phone's keypad, and the voice mail system responds by playing more prompts until the end of the session. This is implemented using a Voice Extensible Markup Language (VXML) browser, also called a voice browser. Think of the voice browser as a web browser that fetches VXML scripts instead of HTML pages over HTTP. VXML is mainly used for voice-based interactive systems such as interactive voice response (IVR) and voice mail. In Cisco UE, the voice browser fetches VXML scripts over HTTP from an internal web server and interprets them to implement the TUI. The Tomcat web server uses Java Server Pages (JSP) to query the voice mail back-end databases to build VXML scripts, depending on the call information and subscriber's mailbox properties.

Sometimes it may be necessary to troubleshoot this aspect of the caller voice mail system interaction, because the caller might hear unexpected prompts or no prompts at all after pressing a digit. When this situation occurs, it is quite possible that the digit pressed on the phone (and translated to a SIP NOTIFY message) did not reach the CRS application or the voice browser software components of the Cisco UE voice mail application.

To understand what is happening if this situation arises, it is useful to troubleshoot the voice browser's operation. It is very difficult to be specific about common problems encountered in this area, so instead this section uses an example of an active TUI session to Cisco UE and discusses the salient aspects of the trace output. Showing how the TUI session should work helps you understand how to troubleshoot any problem that might arise in this area.

You can use the same tracing techniques to troubleshoot any issues with response time for DTMF from the voice mail system. For example, if you feel that the voice mail system's response to DTMF digits pressed on a phone is too slow, you can view the response time in these traces. Look at the time stamp at which the digit arrives at the voice mail application and the time stamp at which the next prompt is played. A large delay between the two time stamps might point to a performance issue on the Cisco UE system.

Example 21-24 shows a voice browser trace for a call forwarded to Cisco UE voice mail. You can use task ID (Task:18000000082, in this case) to differentiate between multiple simultaneous calls. This task ID is generated from the CRS Workflow infrastructure. As discussed in the preceding section, the calling, called, and original called numbers are important for the proper functioning of voice mail. Because voice mail is implemented using a VXML voice browser software component, these Call Information fields have to be passed to the voice browser. As you can see in the voice browser trace in Example 21-24, the voice browser receives all these Call Information fields.

Example 21-24. Start of Voice Browser Traces

```
cue#trace ccn vbroswercore dbug
cue#show trace buffer tail
2702 10/10 12:31:10.338 ACCN VBRW 0 Task:18000000082 Here is the
  DefaultCompilationConfig set in the JVM en_US
2702 10/10 12:31:10.338 ACCN VBRW 0 Task:18000000082 DefaultCompilationConfig will
  be set to en_US
2702 10/10 12:31:10.338 ACCN VBRW 0 Task:18000000082 DefaultCompilationConfig is
  now set to JVM en_US
2702 10/10 12:31:10.341 ACCN VBRW 0 Task:18000000082 enterLevel: SESSION_LEVEL
2702 10/10 12:31:10.341 ACCN VBRW 0 callContact.getSessionID() = 5000000007
2702 10/10 12:31:10.341 ACCN VBRW 0 Task:18000000082 VoiceBrowser sessionid:
  5000000007, taskid: 18000000082
2702 10/10 12:31:10.341 ACCN VBRW 0 callContact.getCallingNumber() = 6003
2702 10/10 12:31:10.341 ACCN VBRW 0 callContact.getDNIS() = null
2702 10/10 12:31:10.341 ACCN VBRW 0 callContact.getDNIS() = null
2702 10/10 12:31:10.341 ACCN VBRW 0 callContact.getCalledNumber() = 5800
2702 10/10 12:31:10.342 ACCN VBRW 0 callContact.getANIIIDigits() = null
2702 10/10 12:31:10.342 ACCN VBRW 0 callContact.getUserToUserInfo() = null
2702 10/10 12:31:10.342 ACCN VBRW 0 callContact.getRDNIS() = null
```

Troubleshooting the Database, LDAP, and Mailbox Activities

This section discusses troubleshooting the Cisco UE voice mail system's interaction with its back-end databases. Here you learn how to interpret SQL database traces, voice mail's interaction with the LDAP database for user information, and message and mailbox activities. These topics are discussed by looking at an example of a subscriber addressing a voice message to another subscriber and sending it. The trace commands needed for this section include

- trace voicemail database all
- trace voicemail ldap all
- trace voicemail message all
- trace voicemail mailbox all
- show trace buffer tail

In the trace segment shown in Example 21-36, a call arrives at the voice mail system with a calling number of 6001. As you can see from the traces, voice mail accesses the LDAP database to resolve the calling number to a user configured on the system to check whether the caller is a valid subscriber on the system or an outside (PSTN) caller calling into the voice mail system. Depending on the outcome of this query, the voice mail system plays different prompts.

In this example, the calling number 6001 is resolved to the user johndoe. Following that action, the system retrieves the user's preferred language (English only in the system used to generate these traces) to select the appropriate prompts to play.

Example 21-36. System Retrieves User Information Based on Calling Extension

```
cue#show trace buffer tail
6082 09/30 18:39:48.614 voicemail ldap "getUserByPhoneNo" 6001
6082 09/30 18:39:48.617 voicemail ldap "getUserByPhoneNo: userDn." /sw/local/
  users/johndoe
6082 09/30 18:39:48.617 voicemail ldap 0 getAttributeValue:
  /sw/local/users/johndoe/Language/preferredLanguage
6082 09/30 18:39:48.618 voicemail ldap 0 getAttributeValue:
  /sw/local/users/johndoe/TelephoneNumbers/primaryExtension
6082 09/30 18:39:48.618 voicemail database 0 Got connection: 2, inUse: 3, active:
  2
```

After the calling user has been identified, the voice mail system verifies that the user has a mailbox, because it is not necessary for every user defined in the system to have an associated voice mailbox.

The SQL statement shown in Example 21-37 is executed and returns the mailbox ID of the given user (for example, the mailbox ID returned here is PERSONAL_00000000000000000000022).

Example 21-37. Retrieving Mailbox Information Based on User Information

```
cue#show trace buffer tail
6082 09/30 18:39:48.619 voicemail database "SQL: " select mailboxid from
  vm_mbxusers where owner=true and userdn='/sw/local/users/johndoe';
6082 09/30 18:39:48.621 voicemail database "Database query results"
```

Troubleshooting the Message Waiting Indicator

The message waiting indicator (MWI) is one of the most important and basic features of any voice mail system. Sometimes, because of many factors, the subscriber does not see the correct MWI changes on the phone. This could be because somehow Cisco CME and Cisco UE got out of sync with each other concerning the MWI status for subscribers. (For example, perhaps Cisco CME was reloaded and it does not keep track of MWI state across reloads.) This section discusses MWI configuration and troubleshooting MWI-related issues.

MWI Operation

With Cisco CME deployments, Cisco UE uses what is called a callout mechanism to change MWI status on the phone. In this mechanism, Cisco CME defines two special extensions: the MWI ON and MWI OFF DNs. These extension definitions have a very specific number format.

For example, the pattern can be 2222.... for the MWI ON DN and 2221.... for the MWI OFF DN, where 2222 and 2221 are prefixes, and the number of dots is equal to the length of the extensions in your Cisco CME dial plan. If Cisco UE must turn on MWI for extension 6001, it places an outgoing call to number 22226001. Cisco CME terminates the call on the MWI ON DN (because the called number matches the dial peer for the MWI ON DN). Cisco CME processes the called number of the call and extracts extension 6001 from it, which is matched by the four dots defined on the MWI ON DN. It then finds all the phones with appearances of extension 6001 on them (there can be several if 6001 is a shared line), and then sends an SCCP MWI ON message to each of the phones. A similar sequence of steps is followed to turn off an MWI lamp by Cisco UE outcalling to the MWI OFF DN. For example, 22216001 is used to turn off MWI for extension 6001.

You can test the operation of your MWI DNs on Cisco CME by dialing the MWI DN prefix followed by the extension from any phone. For example, if you dial 22226001 from any phone, you should see the MWI light activated on phones with extension 6001. This verifies that your MWI DNs are correctly configured and operational.

When you choose the patterns for MWI DNs, keep in mind that such a pattern must not overlap with any other extension or dial peer patterns configured on Cisco CME. Those configurations can interfere with the MWI calls that come from Cisco UE, effectively breaking MWI operation. Also look for any Cisco IOS translation rules that might translate the called numbers for the MWI DN call. This also breaks the MWI operation.

Now that you understand the mechanism between Cisco CME and Cisco UE for turning MWI on or off, the next step is to investigate what happens inside the Cisco UE software components to affect MWI when a voice message is left in or deleted from a mailbox.

When a new message is left in a mailbox, or a subscriber deletes the last new message in the mailbox, the voice mail system has to change the MWI state on the subscriber's phone. However, the voice mail back-end component does not have access to any call control mechanism to place the outcall to Cisco CME. Instead, it depends on the CRS component, which contains a SIP stack for call control. However, there is no direct interface between the voice mail back end and the CRS component.

The CRS component, as discussed in Chapter 3, can start applications on three kinds of triggers:

- E-mail
- HTTP
- Call

To affect MWI changes, the voice mail system (internally) sends an HTTP trigger to the CRS software. A system-level CRS application called ciscomwiapplication, with an associated script called setMWI.aef, exists and is

Troubleshooting Voice Mail VPIM Networking

Chapter 10, "Cisco IPC Express Integrated Voice Mail," explained Cisco UE support for voice mail networking using Voice Profile for Internet Mail (VPIM). It interworks with other Cisco UE systems and Cisco Unity systems in a network. This section discusses some common problems faced in deploying this feature and how to troubleshoot and fix them.

Cannot Send and Receive Network Messages from a Location

If you notice that you cannot send or receive any messages from a particular location, a probable cause is that the local location ID is not configured in the system. Until you configure this on a Cisco UE system, voice mail networking is not enabled. If you are trying to send a network message from a system where a local location ID is not configured, the system plays the prompt "Sorry. The extension you requested is not available." You can verify whether you have configured a local location by using the Cisco UE CLI shown in Example 21-57.

Example 21-57. Displaying Network Locations

```
cue#show network locations
ID         NAME                ABBREV    DOMAIN
303        'Boston'            BOS       cueunity.cisco.com
401        'Bangalore'         BAN       bang.cue.cisco.com
201        'Los Angeles'       LAX       lax.cue.cisco.com

Local location id:
```

In Example 21-57. the local location ID is empty, which indicates that voice mail networking is disabled. You can configure this parameter using the following CLI to set the local location ID to 401:

```
cue(config)#network local location id 401
```

Cannot Send Messages to Cisco Unity

Assume that you have a network with a centralized Cisco Unity at the main campus site, smaller offices with Cisco UEs, and voice mail networking configured between all the sites. You may be able to send messages between the Cisco UE sites and send a message from Cisco Unity to Cisco UE, but you are unable to send a message from Cisco UE to Cisco Unity.

The reason for this problem may be that the sending Cisco UE site is configured in the same domain as that of the Cisco Unity system. The Microsoft Exchange server used by Cisco Unity does not allow other VPIM locations to be in the same domain as itself. For example, the network location configuration shown in Example 21-58.does not work if iptel.cisco.com points to a Cisco Unity server.

Example 21-58. Erroneous DNS Configuration for Cisco Unity

```
cue#show network locations
ID         NAME                ABBREV    DOMAIN
303        'Boston'            BOS       iptel.cisco.com
401        'Bangalore'         BAN       bang.iptel.cisco.com
201        'Los Angeles'       LAX       lax.iptel.cisco.com
```

Separating the domains for the Cisco UE and Cisco Unity systems fixes the problem, as shown in Example 21-59.

Example 21-59. Correct DNS Configuration for Cisco Unity

```
cue#show network locations
```

Summary

This chapter covered voice mail problems. Cisco UE commands to verify the system's voice mail configuration were given, and various GUI-related issues were discussed. The call handoff interface between Cisco CME and the Cisco UE application was covered to illustrate how potential problems in this area may be investigated. The operation of the voice browser, interpreting traces of TUI sessions, and MWI issues were discussed. Numerous examples showed which traces to turn on and how to interpret the fields in the output.

Part V: Appendixes

Appendix A: Cisco IPC Express Features, Releases, and Ordering Information

Appendix B: Sample Cisco UE AA Scripts

Appendix C: Cisco Unity Express Database Schema

Appendix A. Cisco IPC Express Features, Releases, and Ordering Information

The first part of this appendix lists the more common features available in Cisco CallManager Express (CME) 3.2.2 and Cisco Unity Express (UE) 2.1. The second part is a quick guide to ordering a Cisco IPC Express system.

Cisco IPC Express Features and Releases

Highlights of the Cisco CME and UE features are described in this section. For more explanation of the features and how they function, refer to the system administrator guides at Cisco.com:

- Cisco CMEhttp://www.cisco.com/go/ccme
- Cisco UEhttp://www.cisco.com/go/cue

Cisco CME Feature Overview

The Cisco IOS releases corresponding to the more recent versions of Cisco CME are

- 3.012.3(4)T
- 3.112.3(7)T
- 3.212.3.11T
- 3.2.112.3.11XL
- 3.2.212.3.11XL1

The Cisco CME features are divided into the categories of system, call processing and phone, attendant phone, administrative, call coverage, network, and Public Switched Telephone Network (PSTN) trunking. These categories are described in the following sections.

System Features

Cisco CME system features include

- Account codes and call detail record (CDR) field entry
- Station message detail recording (SMDR)/CDR support

Call Processing and Phone Features

Cisco CME call processing and phone features are as follows:

- Cisco CME supports the following Cisco IP phone types:

  Single-line phones7902G, 7905G, 7910G, 7912G

  Wireless 802.11b phone7920

Cisco IPC Express Ordering Information

There are two ways to order Cisco IP Communications (IPC) Express. You can choose one of the bundles, or you can choose a la carte, where you order each item separately:

- Bundles Bundles offer the most popular hardware and software combinations. If you are deploying a new system, several discounted bundles that include both the hardware and software components of Cisco IPC Express are available to make the ordering process easier.
- A la carte If you have a unique configuration, ordering a la carte gives you more flexibility in your choices. It is also the appropriate choice if you already have a Cisco access router for your data infrastructure. To deploy a Cisco IPC Express system, you have to upgrade your Cisco IOS software to a voice-capable image, check that your dynamic random-access memory (DRAM) and Flash capacity on the router are sufficient to support a voice image, purchase a Cisco CME feature license for the desired number of IP phones, and purchase an IP phone seat license for each phone. Cisco UE mailbox licenses can optionally be ordered if you plan on deploying voice mail and an AA application.

Ordering a Cisco IPC Express System a La Carte

Ordering the components of a Cisco IPC Express solution involves several steps:

Step 1.

Select a Cisco IPC platform. Select an appropriate Cisco IPC Express platform for your office. Which one you choose depends on the number of users in your office, the number of IP phones you plan to support, future growth plans, and your budget. The range of available platforms is described in Chapter 1, "Introducing Cisco IPC Express."

Step 2.

Verify memory requirements. Ensure that the platform you select has enough memory to support the solution. Check the memory requirements for the Cisco IOS release you intend to use on Cisco.com.

Step 3.

Select the appropriate Cisco IOS software feature set and release. The following feature sets supported with Cisco IOS Release 12.3(4)T and later are required to deploy Cisco IPC Express: IP Voice, SP (Service Provider) Services, Advanced IP Services, Enterprise Services, and Advanced Enterprise Services. If you want to deploy voice and security features, the minimum feature set required is the Advanced IP Services image.

Next, find the version or release of Cisco CME and Cisco UE (if desired) from Table A-1.

Table A-1. IOS Software Feature Sets and Releases for CME

Minimum Cisco IOS Feature Set	Cisco IOS Software Version	Cisco CME	Cisco UE Hardware
IP Voice	12.3(4)T	Release 3.0	NM-CUE
IP Voice	12.3(7)T	Release 3.1	NM-CUE AIM-CUE
IP Voice	12.3(11)T	Release 3.2	NM-CUE AIM-CUE

Appendix B. Sample Cisco UE AA Scripts

This appendix lists an example of a typical Cisco Unity Express (UE) automated attendant (AA) script. The high-level flow of logic in the script is as follows:

```
If (business hours) then
  Play <menuprompt>
    To enter the phone number of the person you are trying to reach, press 1
    To enter the name of the person you are trying to reach, press 2
  To transfer to the operator, press 0
else
  Play <weekend prompt>
```

The main script is implemented as main.aef, and the dial-by-extension logic (press 1) and dial-by-name logic (press 2) are implemented as the subflowsdialbyextension.aef and dialbyname.aef, respectively. Each of these three scripts is discussed in this appendix.

You can download the binary files (.aef files) for the scripts from the Cisco Press website (http://www.ciscopress.com/title/158705180x). You can view them with the Cisco UE AA Editor, which is a PC application you can download from Cisco.com. Follow these steps:

Step 1. Go to Cisco.com.

Step 2. Search for Software Center.

Step 3. Choose Voice Software.

Step 4. Page down to Cisco Unity Express Software.

Step 5. Choose a software release.

The Cisco UE AA Editor is listed among the files given for a particular release at the Cisco.com Software Center. For example, in the Cisco UE 1.1.2 release listing, the file called CUEEditor1.1.2.exe is the editor. A version of the editor is posted with each release of Cisco UE. Accessing this site requires a Cisco.com login ID. Along with the editor, several more sample AA scripts are posted at this site that you can use to build or customize your own Cisco UE AA.

The Cisco UE AA Editor is discussed in Chapter 9, "Cisco IPC Express Automated Attendant Options." As shown in several of the figures later in this appendix, the script content (steps) is shown in the top-right pane, the tool palette of steps to choose from is in the top-left pane, and the script variables are in the bottom-left pane. Any error or informational messages are in the bottom-right pane.

You construct a script by dragging the steps from the tool palette into the script content pane and filling out the properties for each step. This might include first creating variables and populating their values. To run this script sample on your Cisco UE system, you must record audio .wav files for all the prompts referenced and then upload all the script and prompt files to Cisco UE as described in Chapter 9. It is also important that all variables have values assigned.

The main.aef Script

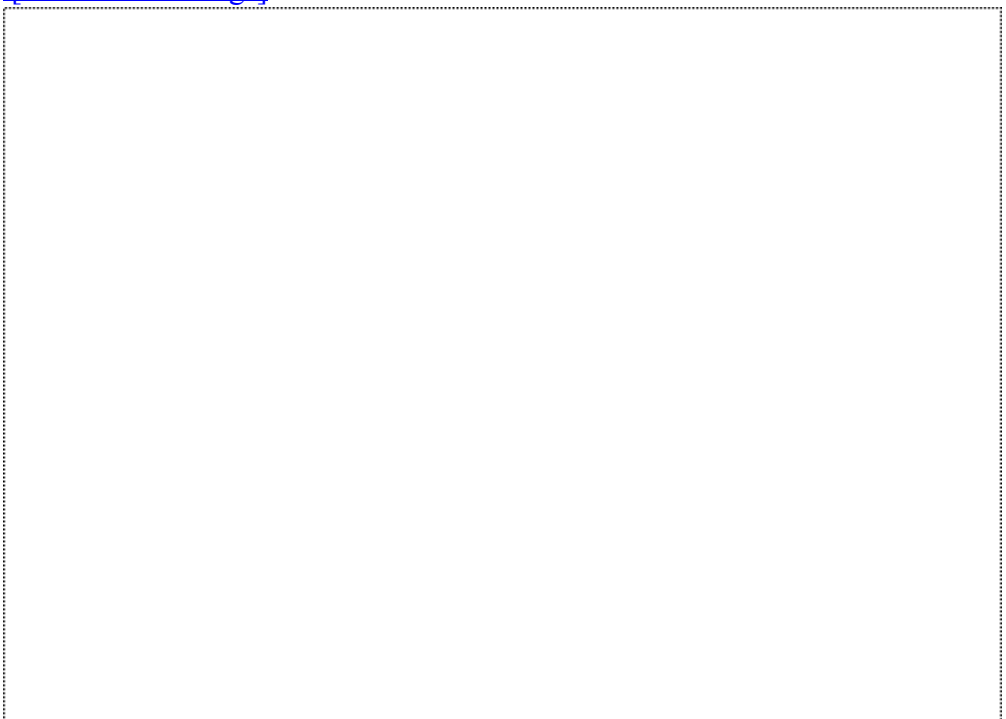
The main.aef script does the following:

- Answers the incoming call (the accept step in the script shown in the next section)
- Checks for business hours and business days (the day of week and time of day steps in the script)
- Provides the normal welcome greeting if it is during business hours (the menu step following the weekdays label)
- Branches to a subflow (a separate script file called dialbyextension.aef) if the caller selects 1 for dial-by-extension (the call subflow to dialbyextension.aef step)
- Branches to a subflow (a separate script file called dialbyname.aef) if the caller selects 2 for dial-by-name (the call subflow to dialbyname.aef step)
- Transfers the caller to the operator (which is a variable) if he or she selects 0 (the Call Redirect step)
- Handles various error conditions

Figure B-1 shows a graphical user interface (GUI) view of the first page of the script, which maps to approximately the top 30 lines of the text representation of the script. Figure B-2 shows the second page, which maps to approximately the last 30 lines of the text representation of the script. The full text and variable definitions of the main.aef script are given in the next section.

Figure B-1. First Page of the main.aef Script

[View full size image]



The dialbyname.aef Script

The dialbyname.aef script allows a caller to spell an employee's name (the name to user step in the script shown in the next section). When a match is found (the get user info step in the script), the caller is transferred (the call redirect step in the script) to that employee's extension.

The full text and variable definitions for the dialbyname.aef script are given in the next section. Figure B-4 shows a GUI view of the main portion of the script.

Figure B-4. The dialbyname.aef Script

[View full size image]

Script Content

Example B-2 is a text representation of the script contents.

Example B-65. The dialbyname.aef Script

```
Start
/* Dial By Name */
Set again = 2
tryagain:
Name To User (contact: --Triggering Contact--, result user:dialname)
  Successful
  Get User Info (user: dialname)
    If ( spokenname == null ) Then
      True
        Create Generated Prompt( spelling type, store in spelledname)
      False
        Create Container Prompt (output prompt: spelledname)
    Implicit Confirmation (contact: --Triggering Contact--)
```

The dialbyextension.aef Script

The dialbyextension.aef script allows a caller to

- Dial the extension (the get digit string step in the script shown in the next section) of the employee he or she wants to contact
- Transfer the call to the selected extension (the call redirect step in the script)

Figure B-6 shows a GUI view of the main portion of the script. The full text and variable definitions for the dialbyextension.aef script are given in the next section.

Figure B-6. The dialbyextension.aef Script

[View full size image]

Script Content

Example B-3 is a text representation of the script contents.

Example B-66. The dialbyextension.aef Script

```
Start
tryagain:
```

Get Digit String(contact: --Triggering Contact--, result digit string: extension)

```
   Successful
   Create Generated Prompt(telephone.number type, store in spelledprompt)
   Implicit Confirmation (contact: --Triggering Contact--)
     No
```

Appendix C. Cisco Unity Express Database Schema

This appendix contains the SQL database schema for Cisco Unity Express (UE). The information in this appendix is relevant to interpreting traces from the voice mail component. It gives the layout of the different tables in the database schema.

Table vm_bcst_heard

```
create table vm_bcst_heard
(
    MessageId           varchar(128)      not null,
    MailboxId           varchar(64)       not null,

    foreign key (MessageId) references vm_message on delete cascade on update
  cascade,
    foreign key (MailboxId) references vm_mailbox on delete cascade on update
  cascade,
    primary key (MessageId, MailboxId)
);
```

Table vm_config

```
create table vm_config
(
    Parameter           varchar(32)       not null,
    Value               varchar(256)      not null,

    primary key (Parameter)
);
```

Table vm_dbversion

```
create table vm_dbversion
(
    dbversion           integer          not null,
    timestamp           date             not null,
    createtime          time             not null,

    primary key (dbversion)
);
```

Table vm_greeting

```
create table vm_greeting
(
    GreetingId          varchar(64)        not null,
    GreetingType        integer            not null,
    MessageLength       integer                    ,
    MessageSize         integer                    ,
    GreetingOid         OID                        ,
    MailboxId           varchar(64)        not null,

    foreign key (MailboxId) references vm_mailbox on delete cascade on update
    cascade,
    primary key (GreetingId)
);
```

Table vm_job

```
create table vm_job
(
    JobId               serial            not null,
    MessageId           varchar(128)      not null,
    Recipients          bytea             not null,
    LocalDomain         varchar(64)              ,
    LocalPhonePrefix    varchar(20)              ,
    HasSpokenName       boolean           not null default false,
    TaskId              bigint            not null default 0,

    primary key (JobId)
);
```

Table vm_mailbox

```
create table vm_mailbox
(
    MailboxId           varchar(64)     not null,
    MailboxType         integer         not null default 0,
    Description         varchar(64)             ,
    MailboxSize         integer                 ,
    MessageSize         integer                 ,
    Tutorial            boolean         not null default true,
    TotalMessageTime    integer                 ,
    MessageExpiryTime   integer                 ,
    Enabled             boolean         not null default true,
    GreetingType        integer         not null default 10,
    OrphanedTime        bigint          not null default 0,
    LastAccessed        bigint                  ,
    ZeroOutNumber       varchar(32)             ,

    primary key (MailboxId)
);
```

Table vm_mbxusers

```
create table vm_mbxusers
(
    MailboxId       varchar(64)       not null,
    UserDN          varchar(256)      not null,
    Owner           boolean           not null default false,

    foreign key (MailboxId) references vm_mailbox on delete cascade on update
    cascade,
    primary key (MailboxId, UserDN, Owner)
);
```

Table vm_message

```
create table vm_message
(
    MessageId           varchar(128)      not null,
    MessageType         integer           not null default 1,
    UseCount            integer           not null default 1,
    Sender              varchar(256)      not null,
    Urgent              boolean           not null default false,
    Private             boolean           not null default false,
    MessageLength       integer                  ,
    MessageSize         integer                  ,
    MessageTime         bigint                   ,
    AttachedMsgId       varchar(128)             ,
    MessageOid          OID                      ,
    UTCTime             varchar(32)              ,
    EndTime             bigint            not null default 9223372036854775807,
    ReceiveTime         bigint            not null default 0,
    NdrReason           integer                        ,
    StartTime           bigint            default 0,
    MWIOn               boolean           not null default false,
    TotalMessageLength  integer           not null default 0,

    foreign key (AttachedMsgId) references vm_message on delete cascade on update
    cascade,
    primary key (MessageId)
);
```

Table vm_usermsg

```
create table vm_usermsg
(
    MailboxId           varchar(64)      not null,
    MessageId           varchar(128)     not null,
    State               integer          not null default 1,
    StoreTime           bigint           not null,

    foreign key (MailboxId) references vm_mailbox on delete cascade on update
    cascade,
    foreign key (MessageId) references vm_message on delete cascade on update
    cascade,
    primary key (MailboxId, MessageId)
);
```

Glossary

AA (automated attendant)

A voice processing system that provides callers with a recorded message and directs them to specific extensions based on their responses to various prompts.

AAA/RADIUS (authentication, authorization, and accounting/Remote Authentication Dial-In User Service) server

Authentication and authorization for users and tracking of connection time provides the framework for access control using database information provided by a RADIUS server.

access control list

See ACL.

ACL (access control list)

A list of devices such as routers, gateways, and servers that are allowed access to certain types of resources in a network.

ANI (Automatic Number Identification)

Provides the calling phone number of an incoming call. In general usage, ANI is the same as CLID.

Asynchronous Transfer Mode

See ATM.

ATM (Asynchronous Transfer Mode)

The standard for which multiple service types (such as voice, video, and data) are organized in fixed-length (53-byte) cells and transmitted over a physical medium such as E3, T3, or SONET. Because ATM cells are easily processed in hardware, faster processing and switching speeds ranging from 155.52 Mbps to 10 Gbps are possible.

authentication, authorization, and accounting/Remote Authentication Dial-In User Service

See AAA/RADIUS.

automated attendant

See AA.

Automatic Number Identification

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